Aalto University

School of Electrical Engineering

Department of Communications and Networking

Joona Mikola

Analysis of Migration Scenarios from Synchronous to Packet Transmission in an Operator Network

Master's Thesis

Helsinki, May 30, 2012

Supervisor: Professor Raimo Kantola

Instructor: Seppo Kuusisto M.Sc

AALTO UNIVERSITY
SCHOOL OF ELECTRICAL ENGINEERING

ABSTRACT OF THE
MASTER'S THESIS

Author: Joona Mikola

Title: Analysis of Migration Scenarios from Synchronous to Packet Transmission in an Operator Network

Date: 30.05.2012 Language: English Number of pages: 8+71

Department of Communications and Networking
Professorship: Networking Technology Code: S-38

Supervisor: Prof. Raimo Kantola
Instructor: M.Sc Seppo Kuusisto

The evolution of telecommunication networks has led to a situation where the usage of traditional fixed telecom services has been replaced with wireless and IP-based solutions. Network operators have identified this trend and have started to migrate their networks towards IP based Next Generation Network.

Network migration is a complicated process and requires a lot of different analyses. Migration needs to be optimized so that the maximum revenue is obtained during a transition process while at the same time customer satisfaction is maintained. This thesis describes how analyses help to manage and predict migration process effectively. Two separate analysis solutions are presented: A tool to predict the development of customer amounts and a tool that helps to obtain the most optimal migration order.

The overall benefits of these tools will become evident in the future when the migration has progressed further but the first obtained results are encouraging. During the implementation of the analyses it was identified that a more evolved analysis platform is needed to replace Microsoft Excel currently in use.

Keywords: TDM, NGN, migration, analysis

AALTO YLIOPISTO
SÄHKÖTEKNIIKAN KORKEAKOULU

DIPLOMITYÖN TIIVISTELMÄ

Tekijä: Joona Mikola Otsikko: Migraatioskenaarioiden analysointi operaattoriverkon migraatiossa synkronisesta pakettipohjaiseen tiedonsiirtoon Päivämäärä: 30.05.2012 Kieli: Englanti Sivumäärä: 8+71
Tietoliikenne- ja Tietoverkkotekniikan laitos Professuuri: Tietoverkkotekniikka Koodi: S-38
Valvoja: Prof. Raimo Kantola Ohjaaja: DI Seppo Kuusisto
Tietoliikenneverkkojen kehitys on johtanut tilanteeseen, jossa kiinteän verkon teleliikennepalvelujen käyttöä on korvattu langattomilla ja IP-pohjaisilla ratkaisuilla. Verkko-operaattorit ovat tunnistaneet tämän kasvavan trendin ja ovat alkaneet muuttamaan verkkojaan IP-pohjaiseen seuraavan polven verkkoon. Verkkomigraatio on monimutkainen prosessi ja se vaatii paljon analyysityötä tuekseen. Migraatio täytyy optimoida niin, että saavutetaan maksimaalinen liikevaihto siirtymävaiheen aikana ja samalla ylläpidetään myös asiakastyytyväisyyttä. Tämä työ tutkii miten analyysejä hyödynnetään migraation hallinnassa ja ennustamisessa. Työssä esitellään kaksi eri esimerkkiä analyyseistä: Analyysiratkaisu, jolla kyetään ennustamaan liittymämäärien muutosta, sekä analyysi, jota hyödynnetään optimaalisen migraatiojärjestyksen määrittelyssä. Näistä analyyseistä saatava kokonaishyöty selviää vasta, kun projekti on edennyt hieman pidemmälle, mutta alustavat tulokset ovat rohkaisevia. Analyysejä implementoitaessa tunnistettiin tarve paremmalla analyysityökalulle tällä hetkellä käytössä olevan Microsoft Excelin tilalle.
Avainsanat: TDM, NGN, migraatio, analyysi

Acknowledgements

I would like to thank my instructor Seppo Kuusisto for the support. In addition, I would like to thank Risto Kuitunen from TeliaSonera for providing necessary information for my thesis. I would also like to thank Raimo Kantola from Aalto University as well as my family for their support during the writing process.

Helsinki 30.5.2012

Joona Mikola

Table of Contents

List of Figures and Tables

List of Abbreviations

AAA	Advanced Access Architecture
ADM	Add/Drop Multiplexers
ANSI	American National Standards Institute
ARPU	Average Revenue per User
AU	Administrative Unit
BRI	Basic Rate Interface
CAPEX	Capital Expenditure
CESoPSN	Structure – Aware Time Division Multiplexed (TDM) Circuit Emulation Service over Packet Switched Network
CLASS	Custom Local Area Signalling Services
CSMA/CD	Carrier Sense Multiple Access/Collision Detection
CWDM	Coarse Wavelength Division Multiplexing
DWDM	Dense Wavelength Division Multiplexing
DXC	Digital Cross Connect
ED	Emulation Device
FDM	Frequency Division Multiplexing
GMPLS	Generalized MPLS
HDTV	High Definition Television
IP	Internet Protocol
IPTV	IP Television
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union Telecommunication Standardization Sector
LAN	Local Area Network
MAN	Metropolitan Area Network
MEF	Metro Ethernet Forum
MPLS	Multiprotocol Label Switching
MSOH	Multiplex Section Overhead
NGN	Next Generation Network

OPEX	Operational Expenditure
OSI	Open Systems Interconnection
PCM	Pulse Code Modulation
PDH	Plesiochronous Digital Hierarchy
POTS	Plain Old Telephone Service
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
PW	Pseudowire
PWE3	Pseudowire Emulation Edge to Edge
QoS	Quality of Service
RSOH	Regeneration Section Overhead
SAToP	Structure-agnostic transport of TDM over Packet
SDH	Synchronous Digital Hierarchy
SIP	Session Initiation Protocol
STM	Synchronous Transport Module
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TDMoP	TDM over Packet
TM	Terminal Multiplexer
UDP	User Datagram Protocol
VC	Virtual Container
VoIP	Voice over IP
VPN	Virtual Private Network
WAN	Wide Area Network
WDM	Wavelength Division Multiplexing

1. Introduction

Telecommunication networks have evolved rapidly during the years. Especially the performance of mobile networks has improved and at the same time bandwidth demands are continuously increasing. Network provider's goal is to maintain necessary revenue growth that is required to sustain operator profitability in an increasingly competitive market environment. In order to achieve these goals the operators need to modernize their networks to enable new services and reduce costs.

Traditionally operators have two fixed network platforms working in parallel. One is a TDM-based circuit-switched legacy network used to provide traditional telecom services like telephony and fax. The other network in use is a packet-switched network used for the Internet. Today it is also possible to deploy telephony services over the packet network. It is easy to understand that maintaining two parallel platforms that offer similar services is not economically sensible. That is why network operators are transforming their networks from TDM to All-IP Next Generation Network (NGN). Shifting to one-platform NGN reduces costs and simplifies network maintenance. The new network makes it also possible to implement more advanced services that will attract customers.

Transformation from TDM to IP can be done in different ways. Some customers naturally adopt new products but usually this natural churn is happening too slowly from the operator's point of view. Some customers must be actively migrated. This process requires extensive customer analysis and interaction which creates additional costs. Network emulation is a technique that is used to simulate the functions of the TDM-network in an IP-network. This can be done with specific emulation devices (ED) so that a connection is migrated to packet without customers noticing any difference. Network emulation requires investments in EDs so it is more economically sensible to deploy it in areas that have more connections.

Determining the suitable migration amounts and methods requires extensive analysis work and a lot of co-operation between the network- and service operator. A network operator has different preferences about migration targets than a service operator. This thesis describes different analyses needed to maintain the most optimal migration process. Mainly the analyses are needed to determine the yearly amounts of active migration and how that affects in emulation amounts and vice versa. In addition to that analyses are needed to determine yearly migration targets. That is also examined in this thesis. The goal is to achieve the most cost efficient migration process while maintaining maximum customer satisfaction.

This thesis is divided in nine chapters. The second chapter focuses on different networks. The main aspects of traditional and next generation networks are presented. The third chapter focuses on the theoretical background. The most essential technologies used in traditional and future IP-networks are also explained. This chapter should be helpful for people who have no background knowledge in telecommunications. Different services provided by networks are examined in the fourth chapter

The reasons and motivation for network migration are examined more thoroughly in the fifth chapter. The chapter describes different migration methods that can be used to optimize the technology change. The sixth chapter describes the different analyses required for migration planning. An overview of migration analysis and planning process is given and different migration strategies are examined. The seventh chapter focuses on an analysis tool used to predict and optimize yearly migration amounts while the sixth chapter describes an analysis process of how yearly migration targets are chosen.

Another case example is presented in the eight chapter: an example of how yearly migration targets are determined. The ninth and the final chapter shortly summarize the thesis. Conclusions about the usefulness of analyses and how the migration could be improved are given.

2. Network overview

The second chapter focuses on different kinds of networks. The differences between circuit and packet switched networks are examined. The characteristics of the circuit-switched legacy and packet-switched Next generation networks are also examined. This chapter provides a useful insight of the environment network migration functions in.

2.1 Circuit-switched Networks vs. Packet-switched Networks

Traditional telephone networks are circuit switched. In circuit switching a dedicated channel is reserved for each telephone call. This channel remains open and active during the whole call and it cannot be used by any other data or phone calls. Usually the calls are routed through several switches that hold switching state for the call. The entire data is routed along the same path. The dedicated circuit offers several advantages. There is no interference, connections have a low delay and there is no need for channel sharing. The disadvantage of circuit switching is that it is not very efficient for short flows or bursty traffic. For example during a telephone call there are some silent moments when neither person is speaking. During this only a very small amount or no useful data is transmitted along the circuit. The resources remain still reserved even though no data is sent which leads to a less than optimal operation.

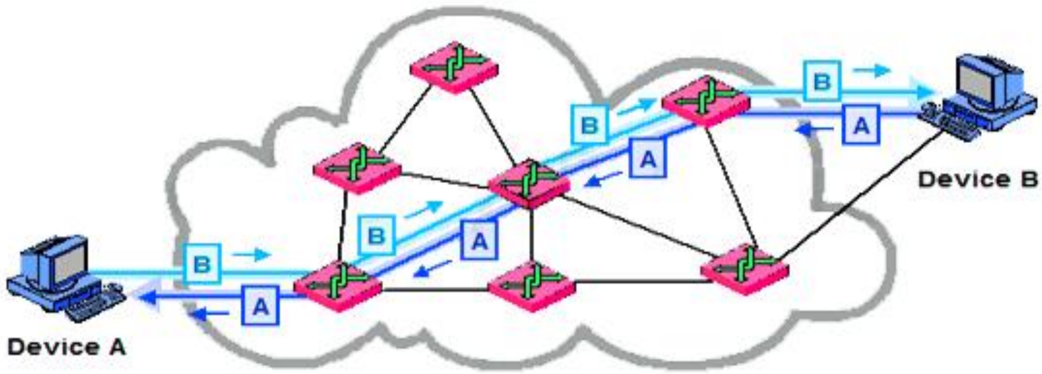

Figure 1 Circuit-Switched Network

.

In packet switching the data is broken into small packets that are sent into a network. These packets travel in the network trying to find the best possible route to the destination. When sending a packet a specific header containing information about its destination is added to the packet. This header may also have sequence numbers and information about how many packets were sent. This information enables the destination side to put the packets in the correct order and to find out if packets are missing. If a packet doesn't reach its destination the destination host can request the missing packet to be resent.

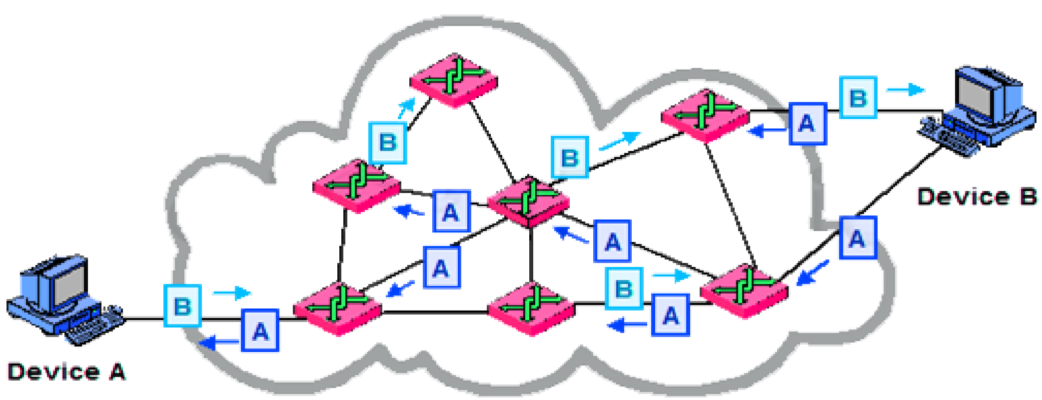

Figure 2 Packet-Switched Network

Packet switching allows better utilization of bandwidth compared to circuit switching. For most of the traffic there is no need for reserved channels. Data is sent to a network when there is something to send and during the silent periods the bandwidth is available to other users. Packet switched networks are also cheaper and easier to expand. The disadvantages of packet switching are potential packet losses and increased delays when there are lots of users in the network. Because of the possibility of packets arriving out of order, packet switching is not a very suitable solution for some data streams like real-time video. [1]

2.2 Public Switched Telephone Network

Traditional telephone networks are circuit switched. Telephone networks that have public access are generally called Public Switched Telephone Networks (PSTN). The PSTN consist of copper wires and optical fibres interconnected with different switches and exchanges. At the beginning of the PSTN these switches were manually operated but today these manual switches have been replaced by automatic electronic switches. PSTN is a global network which is divided to smaller networks managed by different operators. These networks need to interconnect so that their subscribers can call to subscribers using other networks. Traditionally the PSTNs are based on TDM technology and use digital signalling. PCM is the method used in converting an analogue signal to digital format. TDM and PCM are examined in the next chapter. Current signalling technology in use is called Signalling System No. 7 (SS7). SS7 is an out-of-band signalling method which enables the implementing of more advanced services.

PSTN’s main function is to switch voice calls. It is not very suitable for data transmission because data has different characteristics compared to voice. Data has, for example, a variable use of the bandwidth and the need for higher transmission speeds. PSTN has also issues concerning its flexibility. It is built on an infrastructure whereby only the vendors of the equipment develop the applications for said equipment. At the same time deregulation has increased competition and that has encouraged operators to develop new services and applications. For that purpose a more open infrastructure, by which many vendors can provide and develop applications is needed. [2]

2.3 Integrated Services Digital Network

Integrated Services Digital Network (ISDN) is a design for a completely digital network. While PSTN uses digitalized switches, it doesn't offer the end-to-end digital connections provided by ISDN. ISDN is a circuit-switched network system that mainly uses the same switches and exchanges with PSTN. ISDN can also provide access to packet networks. The main advantage of ISDN is its ability to handle different types of information, like data, audio and video. It also provides a single interface for all devices, such as telephones, fax machines and computers.

ISDN has two different user interfaces:

- Basic Rate Interface (BRI)
- Primary Rate Interface (PRI)

BRI is a 144kbps service that is separated in three channels. There are two 64kbps bearer channels, also known as B-channels, and one 16kbps data channel, also known as D-channel. The B-channels are used to transmit the actual data, while the confusingly named D-channel transmits different signalling and control information. The PRI is transmitted over an E1-carrier that has the 2048kbps transmission rate. In ISDN PRI this 2048kbps channel is divided to 30 64kbps B-channels, one 64kbps D-channel and one 64kbps channel used for timing and alarms. BRI is the most appropriate for individual use and for small businesses, while PRI is mostly used only by businesses.

Generally ISDN can be considered to be a more evolved version of PSTN. The end-to-end digital connection enables better quality and higher transmission speeds. One of the ISDNs biggest attractions was its data transmission capabilities, which were more evolved, compared to those of PSTN. For example, ISDN offered higher data rates and made it possible to access Internet while the telephone was in use. Today in data transmission the ISDN has been mostly superseded by broadband Internet that offers much higher rates and lower prices. [3]

2.4 Next Generation Network

The Next Generation Network (NGN) is a quite broad term. It is used to describe the architectural change in telecommunications networks and it consists of multiple technologies and protocols. ITU-T has created several documents where NGN characteristics are described. The fundamental principles of the NGN are documented in ITU-T recommendation Y.2001 [4]

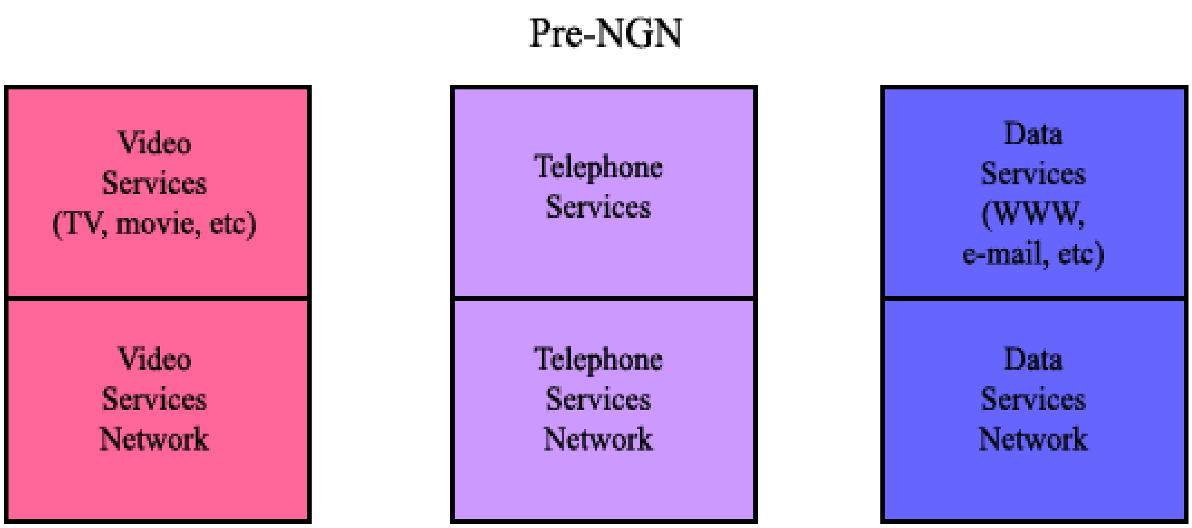

Figure 3 Traditional networks

NGN is a packet-based network, so it is ideal for data transmission. The main motivation for NGN is the convergence of different services and networks. Data, voice and video can all be transmitted in the same network. In NGN the service and transport levels are separated, which means that the services are independent of transport details. This enables the service provider to implement new services simply by defining them in the service layer, without consideration for the underlying transport layer.

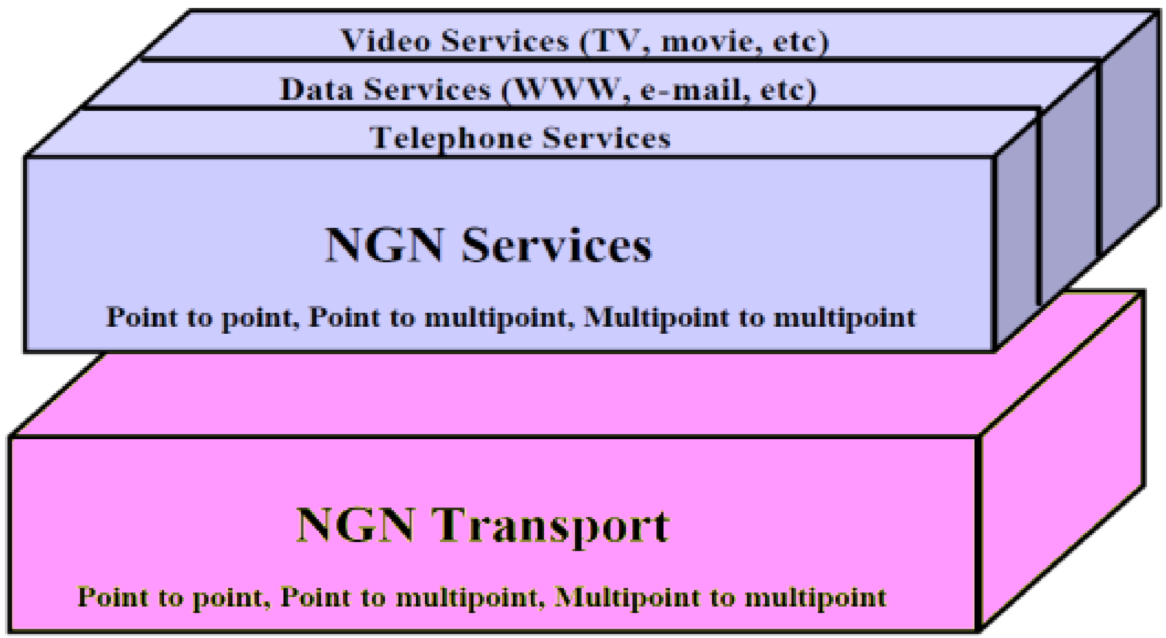

Figure 4 NGN Layers

Additional flexibility is obtained by the ability to use the services provided by NGN from different access networks. One key requirement of NGN is to provide broadband capabilities with end-to-end QoS and transparency. NGN must also be able to support different legacy networks. This can be achieved with emulation, for example. Emulation is discussed later on in this thesis.

Commonly the NGNs are built around the IP-protocol and that is why the term “all IP-network” is often used around NGN. IP is the widely accepted standard for which most of the new applications are built. This enables easier integration and interoperability between the applications within networks. [5]

3. Technologies used in Networks

There are different networks and many different technologies are used in them. This chapter focuses on these different network technologies. The services provided in the network depend heavily on underlying technology so that is why it is important to have understanding about that. This chapter provides about both the technologies working on older platform and newer technologies used in the NGN.

3.1 Time Division Multiplexing

In circuit switched networks multiple transmissions need to be transferred along the same transmission medium. Time Division Multiplexing (TDM) is used in circuit switched networks to achieve this. TDM is a technique where the time domain is divided in slots and these slots are allocated to different sub-channels. TDM allows multiple users to transmit data on the same transmission medium. During the time slot the full bandwidth of the channel is reserved to the sub-channel occupying the slot. One TDM frame consists of one timeslot per sub-channel plus synchronization and signalling channels. TDM is widely deployed in traditional PSTN transmission protocols, PDH and SDH, which are discussed later in this chapter.

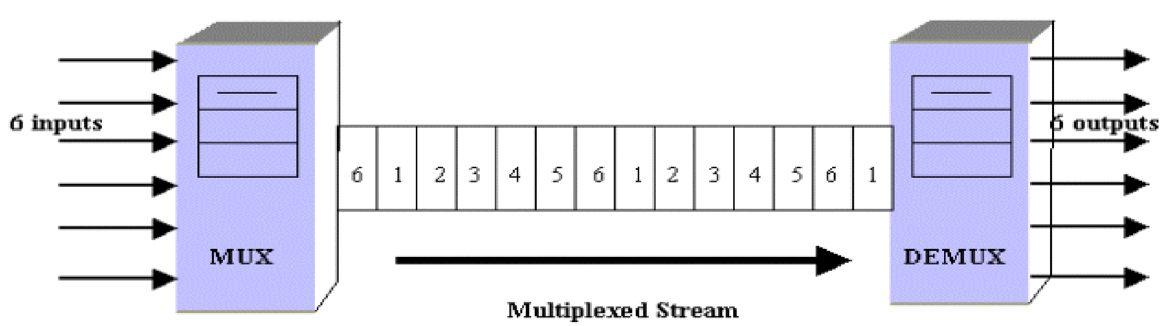

Figure 5 TDM system

The basic idea of TDM is presented in figure 5. Six different channels arrive to the multiplexer where they are buffered. The buffer length is equal to the length of one time slot. These buffers are then sequentially scanned so that a multiplexed data stream is formed. The de-multiplexer receives the data stream, separates the data back to their channels and outputs it to the correct lines.

TDM is mainly used in PSTN to multiplex digital signals. Pulse Code Modulation (PCM) is a method where analogue signal is coded in digital format. The main idea of the PCM is to sample the analogue signal at regular intervals and then quantize values to the nearest digital value. In telephony each sample is represented with 8 bits, so there are $2^8 = 256$ possible quantization values.

In order to obtain the necessary quality for the signal, samples must be made frequently enough. The required sampling rate for telephone calls can be derived from Nyquist Sampling Law [6], which states that the minimum sampling rate should be twice the maximum frequency of the signal so that the full information in the signal can be preserved. In telephony the voice signal range is between 300-3400Hz, so the minimum sampling rate should be 6800Hz, but for practical reasons an 8000Hz sampling rate is used.

When the signals have been converted to digital format, TDM can be used to obtain larger aggregate data streams. Currently the International Telecommunications Union (ITU) has two standardized versions of PCM multiplexing [7]:

- The 30-channel E-carrier, which is used in Europe, Asia and on international links
- The 24-channel T-carrier, which is used in America and Japan.

Here we focus more on the European version. In the 30-channel multiplexing standard, the transmission channel is represented as a time frame split in 32 time slots. The timeslots are numbered from 0 to 31. The 8000Hz sampling rate means that the samples are taken every

125µs. That is also the size of the TDM time frame, while the size of the time slot is 125µs / 32 = 3.9µs.

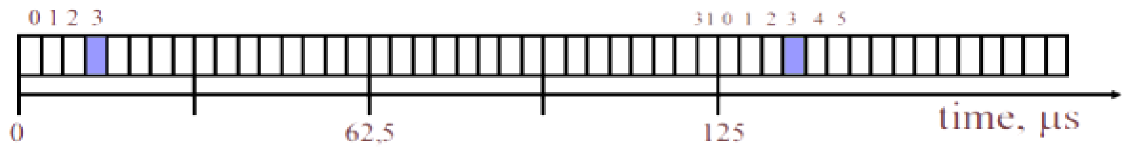

Figure 6 PCM system

Only 30 channels from the available 32 are used to transmit speech. The channel occupying the time slot 0 (TS0) is used to indicate the start of the frame. At the sending end, a special 8-bit pattern called frame alignment pattern is inserted into the TS0. This pattern is used to identify the start of the frame. At the receiving end two-frames worth of bits are picked and the first 8 bits are checked. If the frame alignment pattern is not detected, the inspected area is shifted by one bit and the check is redone. This process is repeated until the frame alignment pattern is recognized. Traditionally the channel in TS16 was used to transmit signalling information related to call control. That left channels 1-15 and 17-31 to be used in speech transmission. If each sample is represented with 8 bits and the sampling rate is 8 kHz, the transmission speed of a single slot is:

8 kHz * 8 bit = 64kbps

From the equation above the transmission speed C of the whole system of 32 slots can easily be calculated:

C = 32 * 64kbps = 2048kbps.

The more popular form for C is simply to abbreviate it to ‘2Mbps’. Traditionally circuit switched digital telecommunications networks are built on these 2Mbps connections. In addition to speech they can be used to transmit data.

TDM has been the leading multiplexing technology for about 30 years. Before TDM, Frequency Division Multiplexing (FDM) was dominant. The idea of FDM is to divide the bandwidth available into smaller parts. Each transmitting signal was attached to a certain part of the bandwidth so that multiple users could deploy the same transmission medium. FDM was eventually largely replaced by TDM systems that had better support for data and digital transmission. [8-9]

3.2 Plesiochronous Digital Hierarchy

Plesiochronous Digital Hierarchy (PDH) was the first internationally standardised form of digital higher-order multiplexing. There are both European and American standards for PDH but here we focus on the European version. The word plesiochorous comes from the Greek language and roughly translates to “almost synchronous”. PDH has been mostly replaced, especially in core networks, by SDH and other more advanced technologies, but in access networks there are still noticeable amounts of PDH-devices. In Europe PDH-systems are based on the 30-channel PCM-multiplexing standard discussed earlier. The basic transfer rate of PDH is therefore 2Mbps, with 30 64kbps channels used to transmit speech and two 64kbps channels for synchronization and signalling. Alternatively the bandwidth can be used for other purposes, for example data transfer. This first level PDH hierarchy is known as E1. Different PDH hierarchy levels are presented in Table 1.

Table 1 PDH Hierarchy levels and Data Rates

Class	No. of 64 Kbit/s Channels	Actual capacity Mbit/s	Nominal Capacity Mbit/s
E1	30	2.048	2
E2	120	8.448	8
E3	480	34.368	34
E4	1920	139.264	140
E5	7680	564.148	565

As can be seen in Table 1, there are five different PDH hierarchy levels, each with roughly 4 times higher transmission rate than the previous level. So four 2Mbps E1s are multiplexed to form the E2 channel with a transmission rate of 8.448Mbps, four E2s for the E3 with a transmission rate of roughly 34Mbps and so on.

PDH's almost synchronous nature means that different parts of the PDH system are operating on slightly varying rates. That leads to a need to add justification and stuffing bits for each multiplexing session. For example if we simply multiplex four 2.048Mbps E1s to E2, the E2 transmission rate should be 4 * 2.048Mbps = 8.192Mbps. The deviation from the calculated E2 capacity (8.448Mbps) results from the addition of stuffing and justification bits.

PDH's multiplexing and hierarchy levels make it a rather cumbersome technology. The existence of justification bits requires a step-by-step de-multiplexing process within the PDH-systems. For example, in order to extract a 2Mbps E1 block from the 140Mbps E4, each stage of de-multiplexing must be performed. At first E4 must be de-multiplexed to four E3s, then these to E2s, and finally E2s to E1s. It is easy to understand that this is quite an inflexible solution, which requires a huge number of devices.

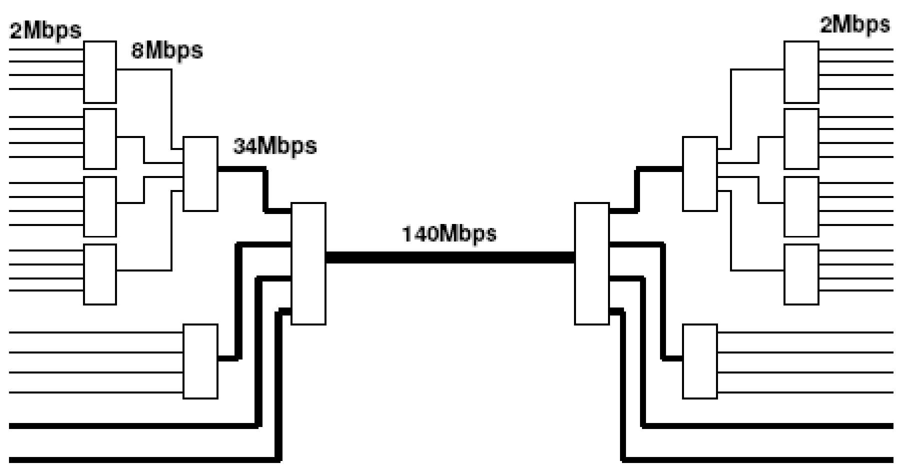

Figure 7 PDH multiplexing and de-multiplexing

Another problem is that PDH doesn't have a standardised control mechanism, which means that it can differ between manufacturers. There are some spare overhead bits that are being used for management, but they have limited bandwidth and are hard to locate in a 140 Mbps stream without the cumbersome de-multiplexing. Optical interfaces are also not standardized in PDH. These are just a couple of reasons why the more flexible SDH technology was developed. [10]

3.3 Synchronous Digital Hierarchy

The growth of network traffic and problems with PDH lead to the need for developing a new transmission technology. For this purpose the Synchronous Digital Hierarchy (SDH) was created. SHD was standardised by ITU and it is used globally, excluding North America. In North America a technology called SONET, which is quite similar to SDH, is used. Actually the American National Standards Institute (ANSI) developed SONET before SDH at the beginning of the 1980's. SDH, which is strongly based on SONET but adapted to European networks, was developed by ITU-T by the end of the 1980's.

The transmission data streams of SDH are called Synchronous Transport Modules (STM). The first SDH hierarchy level is called STM-1 and its transmission rate is 155.52Mbps. SDH hierarchy levels and their transmission rates are presented in Table 2. From the table it can easily be seen that the transmission rates of each SDH level are exact multiples of STM-1's 155.52Mbps data rate.

Table 2 SDH hierarchy levels and data rates

Class	Actual Capacity Mbit/s
STM-1	155.52
STM-4	622.08
STM-16	2488.32
STM-64	9953.28

3.3.1 STM-1 Frame

The main transport element of the SDH networks is the STM-1, so it is good to examine it more closely.

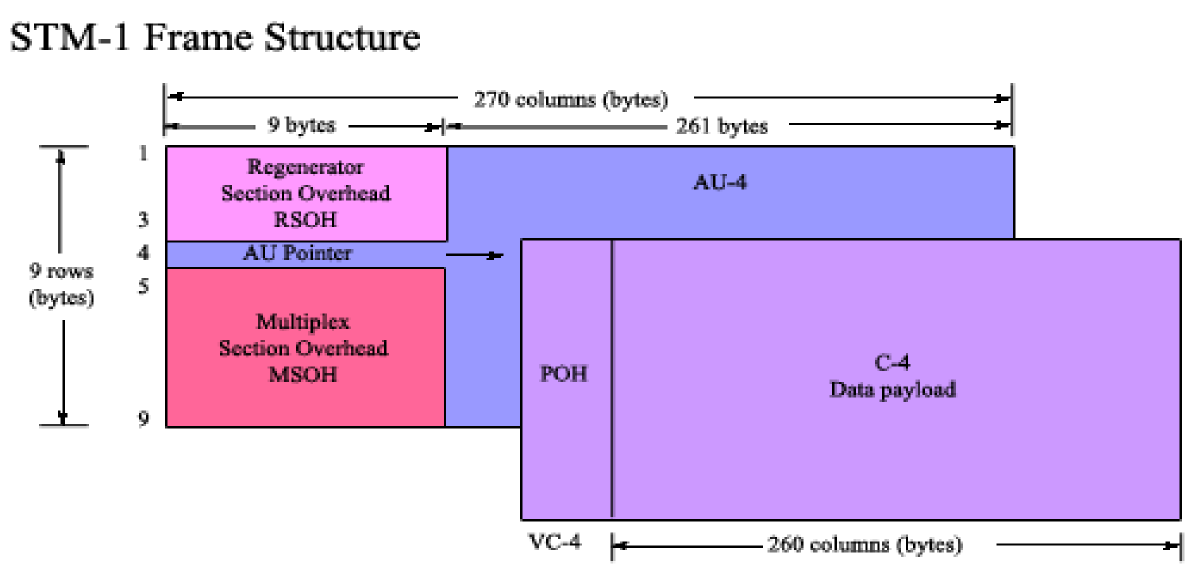

Figure 8 STM-1 Frame Structure

The structure of the STM-1 frame can be seen in Figure 8. The STM-1 frame is a matrix with 9 rows and 270 columns of bytes. Each frame is repeated 8000 times in a second, so the transmission rate is:

(9*270*8) bits * 8000/s = 155.52Mbps.

The SDH network must be capable of transmitting PDH data, so for that reason specific containers have been defined in the SDH standard that can carry this data. PDH streams from E1 to E4 are synchronized and then packed in these containers. The path overhead (POH) which contains control and supervisory information, is added to the beginning of the container. The sum of the container and POH is called a virtual container (VC). VC-4 is used for 140Mbps E4 and VC-3 for E3, while VC-12 is used for 2 Mbps E1. VCs can be packed into larger VCs so that VC-4 can consist of three VC-3s or 63 VC-12s. From Figure 8 the payload of STM-1 frame can be calculated:

(9*260*8) * 8000 = 149.76Mbps

The result indicates that STM-1 frame can carry one VC-4 or three VC-3s. In addition to payload, the STM-1 frame has two main fields: AU (administrative unit) pointer and section overhead (SOH). AU pointer is used to specify where the payload starts. SOH has two fields: multiplexer section overhead (MSOH) and repeater section overhead RSOH. Both contain different control and frame synchronization information. RSOH is used by all network elements while MSOH is accessible to every element, other than regenerators.

SDH network is divided into sections and paths. The physical network is divided into two sections called repeater section (RS) and multiplexing section (MS), while the logical network is divided in lower and higher-order paths. The section indicates the distance between two network elements and the path tells us the distance between the points where VC is formed and terminated. If VC is formed by multiplexing smaller VCs, it corresponds to a higher-order path and if VC carries non-multiplexed flow it belongs to a lower-order path.

3.3.2 SDH Network Elements

Different kinds of network elements are used in SDH networks. These are regenerators, terminal- and add/drop multiplexers (ADM) and digital cross connects. Different systems make the SDH much more flexible than the PDH. Terminal multiplexers (TM) are located at the end points of the SDH network and are used to multiplex and de-multiplex PDH and SDH streams from STM-n frames. For example with TM, 63 E1 streams can be extracted from one STM-1 frame. An ADM is a multiplexer that can add or drop single streams from STM-n.

Figure 9 Terminal- and Add/Drop multiplexers

Digital cross-connect (DXC) devices are used to rearrange different SDH connections. They can transmit information within the SDH network and different lower-level bit streams can be attached straight to them. With DXC the connection is set up and released by the network operator. Generally DXCs are the largest and the most expensive SDH-elements.

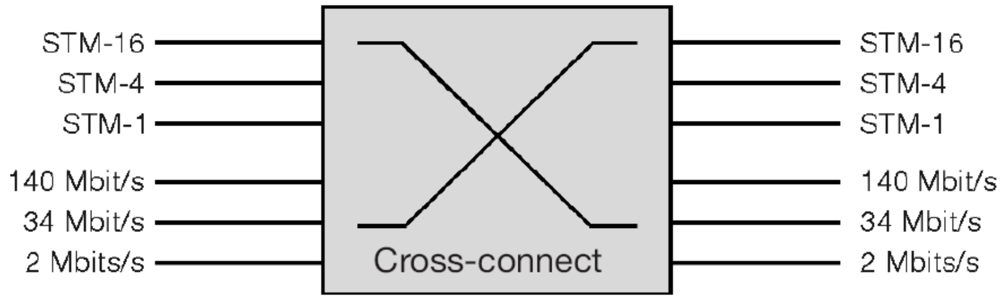

Figure 10 SDH Digital Cross-connect

Regenerators are the least complicated elements and they are simply used to regenerate the line signal in order to maintain acceptable signal strength. [8,10]

3.4 TCP/IP

The Internet Protocol Suite, more commonly known as TCP/IP is a set of different protocols, applications and network media used in the Internet and similar networks. It is the most commonly used protocol in networks today and also the majority of the traffic in next generation networks is based on TCP/IP. Two of the most popular protocols of the set are Transmission Control Protocol (TCP) and Internet Protocol (IP), hence the name TCP/IP. Other protocols of this suite are for example ICMP, ARP and UDP, while the example applications are TELNET and FTP. The Open Systems Interconnection model (OSI model) is another way to describe network technologies and like OSI model, TCP/IP also uses different layers of functionality. The layer architectures of OSI and TCP/IP model can be seen in Figure 11.

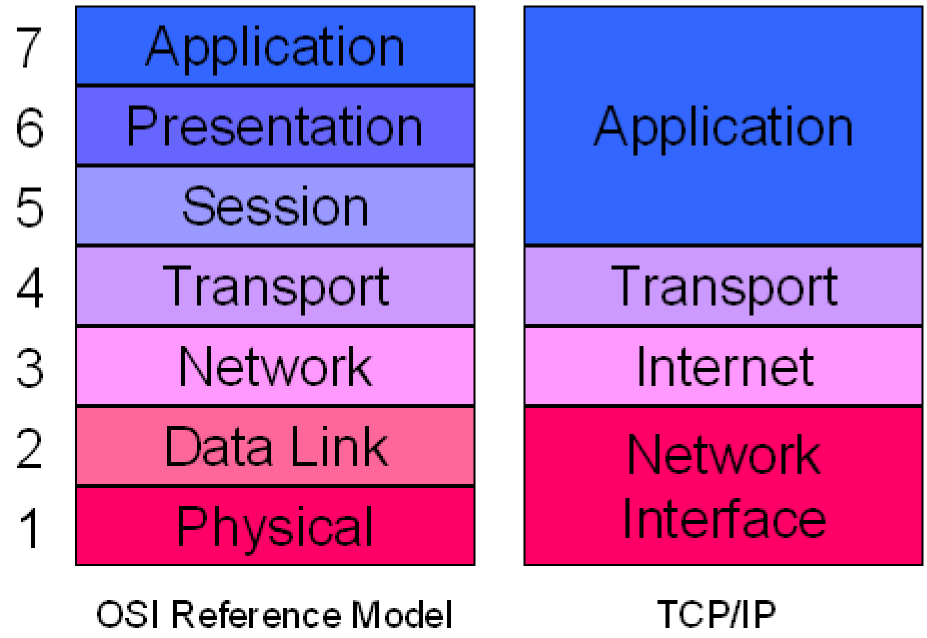

Figure 11 OSI and TCP/IP reference models

As can be seen in Figure 11, the TCP/IP model has four layers. The application layer contains direct interaction with programs and it includes commonly known protocols such as HTTP for web browsing and SMTP for e-mail. The transport layer is responsible of transporting data between the application layer and the Internet. The most common transport layer protocols are TCP and UDP. On the Internet layer IP-protocol receives packets from the transport layer, adds data header including sending and receiving computer addresses and passes these datagrams to the network interface. The network interface then sends these packets over the network. Today the most important network interface layer protocol is called Ethernet, which will be discussed later in this thesis. The basic idea of the TCP/IP architecture is that the data is packed in packets and each packet must be processed by each layer. At each processing step a different layer header containing different control and routing information is added to the packet. This is called packet encapsulation. Figure 12 describes a typical encapsulation case in TCP/IP, where data IP packet is sent to the Ethernet network using TCP.

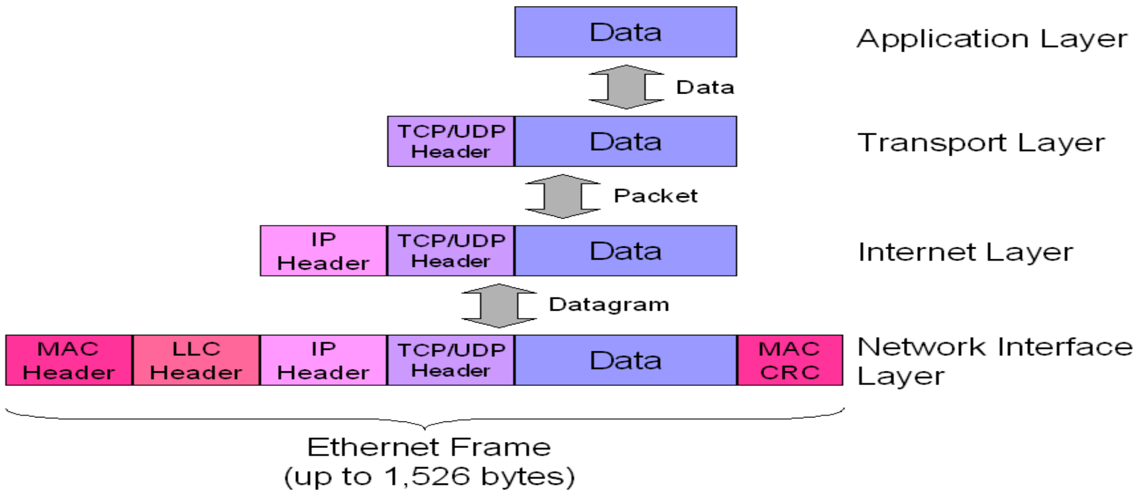

Figure 12 IP packet encapsulation

The main purpose of the IP-protocol is to transfer packets from sources to destinations. Generally IP-packets are called datagrams. The IP defines the addressing methods and structures for datagram encapsulation. The sources and destinations are identified with specified binary addresses called IP-addresses. The first major version of IP was IP version 4 (IPv4). IPV4 defined addresses that were 32 bits long. This offered an address space of 2^{32} = 4,294,967,296 addresses. An increased demand in IP-based services has led to a situation where there simply aren't enough IPv4 addresses for everyone. For this purpose IPv6, that has an address space of $2^{128} = 3.4^{1038}$, was developed. IP is a connectionless protocol, which means that the communication between hosts occurs without any handshaking procedure. Basically the host can send packets to the destination without being sure that the sender is prepared to receive them. This combined with the unpredictable routes of packet network means that the IP cannot guarantee that the packets will arrive to their destination. [11]

TCP is used to provide highly reliable transmission between hosts in packet switched networks. TCP data is sent in segments that are encapsulated in IP datagrams. TCP uses three-way handshaking to form a connection between hosts [12]. First the sender sends a SYN packet to the receiver, who acknowledges this by replying with a SYN/ACK packet. Finally the sender replies to this with an ACK packet and thus the connection is established. Connection can be terminated in a few ways, but a similar three-way handshaking method is considered to be most common. TCP makes sure that the data is received in the right order by attaching a sequence number to each transmitted octet. Damaged segments can be identified with a checksum that has been added to them. In the original version of TCP, reliability is achieved by requesting an ACK packet from the receiver for every packet sent. If this acknowledgment is not received within the timeout interval, the data is retransmitted. This can lead to an inefficient performance when multiple packets are lost from one window of data. In the cumulative acknowledgement, implemented in the original version, only a single lost packet in a window can be identified per round trip time. TCP Selective Acknowledgment Options that was introduced in 1996 is used to counter this problem. It allows the receiving end to acknowledge correctly received discontinuous block of packets. Flow Control and Congestion Control are other TCP mechanisms used to counter the problems of cumulative acknowledgements. [12-13]

UDP is a transport layer protocol, which enables applications to send data to each other without any communications set-ups or path reservations. That means that UDP does not provide the reliable transport TCP does. For that reason UDP is also much more of a light-weight protocol. It is suitable for use for example in transmitting voice in an IP network, because the three-way handshaking used in TCP would cause delays that would then hinder voice communication. [14]

3.5 Metro Ethernet

Metro Ethernet is an Ethernet based network that covers metropolitan areas. More generally Metro Ethernet is commonly used in access networks, so basically they are built between core networks and customer premises. It can also be used to bridge or connect to separate enterprise Local Area Networks (LANs).

Ethernet was developed in the beginning of the 1970’s and it is one of the first packet-based transmission technologies. Ethernet includes many different standards and it is constantly evolving. Traditional Ethernet was based on idea where computers attached to network would use the same transmission medium. That enabled multiple users to send packets at the same time, and because of the shared medium, that lead to packet collisions. There is a technique called Carrier Sense Multiple Access/Collision Detect (CSMA/CD) that is used to handle these collisions appropriately. Today the Ethernet standards support full-duplex, which means that a network node can transmit and receive data simultaneously. Full-duplex and different switches have made CSMA/CD obsolete. Until recently the highest data rate supported by the Ethernet was 10 Gbit/s, but today the possible transmission rates of the Ethernet are 40 Gbit/s and 100 Gbit/s, which are much higher rates compared to the ones provided by traditional TDM technologies. The Ethernet's lower prices and a larger number of vendors have also made it the dominant LAN technology over Token Ring [15-16]

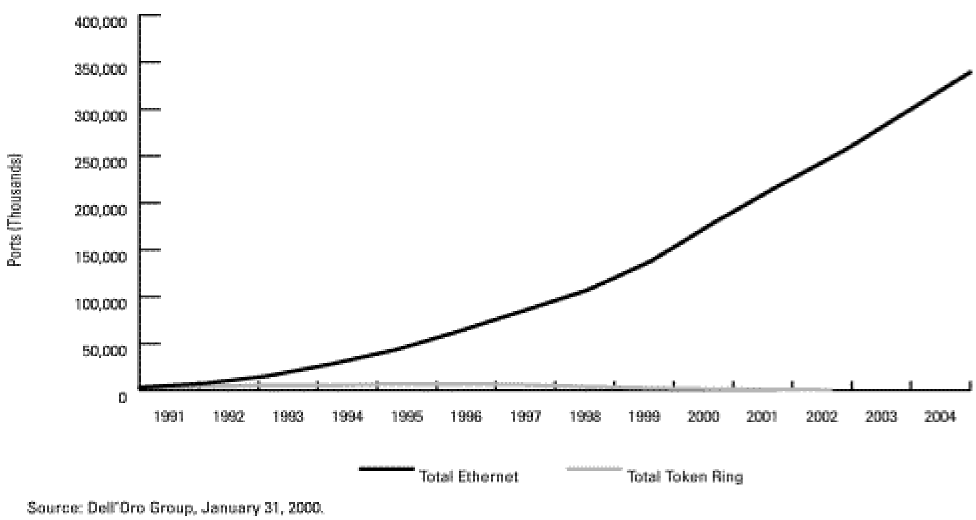

Figure 13 The Growth of Ethernet Ports

Metro Ethernet Forum (MEF) is a non-profit organization that is dedicated to accelerating the adoption of carrier-class Ethernet networks and services. The main idea of a Carrier Ethernet Service is that the network operator can provide Ethernet for a customer as a service. Traditional Ethernet was designed for LANs, so Carrier Ethernet can be considered to be an extension to Ethernet that is deployed in wide area networks (WANs) and in metropolitan area networks (MANs). The MEF has defined service attributes and parameters for successful implementation of Ethernet services in WANs and MANs. Three different types of services that can be delivered through Metro Ethernet have been identified [17]:

- The point-to-point service called E-Line
- The multipoint-to-multipoint service called E-Lan
- The Rooted-Multipoint Ethernet Virtual Connection for multicast domains called E-TREE

The MEF does not actually create new standards but it supplies different white papers, case studies and technical specifications that are used to leverage current standards and define new ones. Figure 14 describes normal Metro Ethernet network architecture. 10 Gigabit Metro Ethernet ring is used to connect different customer networks into a core IP Backbone. Customers like different businesses can have their own local area Ethernet rings that are also connected to Metro Ethernet.

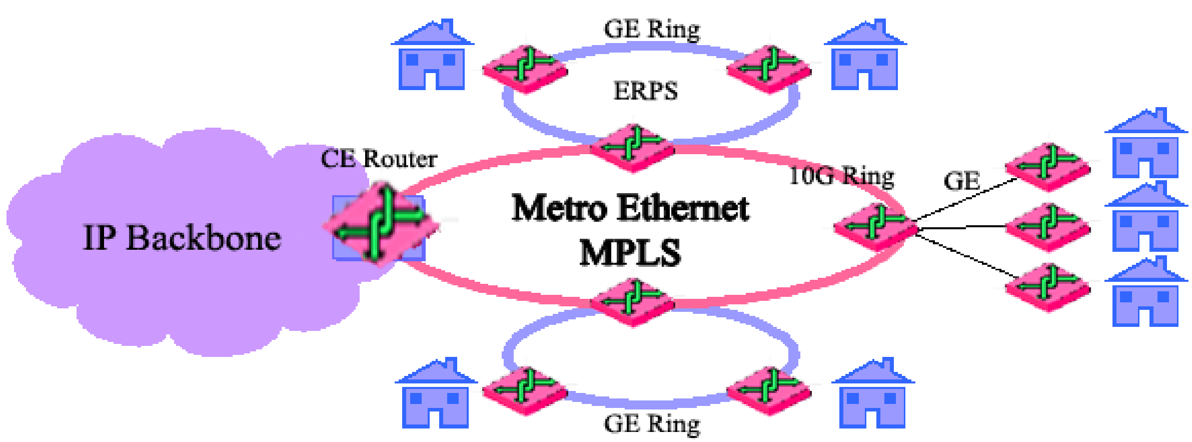

Figure 14 Metro Ethernet Network

Traditionally TDM-based technologies like SDH and PDH have been used in metropolitan area networks. They are replaced by Ethernet because it offers much lower operational and capital expenditures (OPEX and CAPEX). One important reason is also that packet based traffic has overtaken all other traffic types. The main problems in Metro Ethernet networks are that they are not as reliable as SDH networks and they can't offer as good end-to-end Quality of Service (QoS) guarantees. To target these issues Metro Ethernet usually uses IP and Multiprotocol Label Switching (MPLS)

The main function of MPLS is to route packets in networks. It supports many different transmission protocols like ATM, Frame Relay and IP. In MPLS a label is attached to the packets. This label has information of the next destination router of the packet, and when a packet arrives to the router a new label indicating the next destination is added to replace the old one. The original motivation behind MPLS was to make routers faster. It was observed that the usage of labels enables routers to make routing decisions at a much faster rate, because they only have to analyse the next destination of the packet, rather than perform a complex route lookup based on the destination IP address. Later this advantage has been found to be rather marginal.

MPLS offers advanced traffic engineering capabilities. Labels can have different priority levels, which makes SDH-like QoS guarantees possible. This enables managing traffic characteristics, such as who can send data, where to and what kind of data can be sent. More importantly MPLS is totally independent of different link- and network technologies and that allows the integration of networks with different technologies. In Metro Ethernet different services have to be provisioned and monitored over different kinds of data and switches. Generalized MPLS (GMPLS) is an extension of MPLS that works as a control plane in Metro Ethernet managing mixed data and switches. [18, 26]

3.6 Wavelength Division Multiplexing

When traffic in the networks started to increase, new techniques to support higher transmission capacities were needed. Wavelength Division Multiplexing (WDM) is a technique that allows a number of channels to be sent on a single optical fibre by using different wavelengths. WDM allows both uni- and bi-directional transmission. One downside of WDM is that it is end-to-end technology, which means that if the fibre capacity is wanted to be utilized in the middle of the fibre, WDM devices need to be installed there. That makes it quite expensive to implement WDM in rural regions.

There are two different versions of WDM: Coarse Wavelength Division Multiplexing (CWDM) and Dense Wavelength Division Multiplexing (DWDM). CWDM was originally developed during the 1980’s but it has been modified since then. Currently CWDM has a channel spacing of 20nm and it uses wavelengths from 1270nm to 1610nm. DWDM was developed in the early 1990’s. The channel spacing in DWDM can be as small as 0,8nm and it operates around 1550nm band. DWDM can transmit over 100 different channels on one fibre, while with CWDM there are approximately 18 channels available. DWDM has a much higher capacity and a bigger range than CWDM, but it is also more expensive and requires more complex technology. For these reasons the use of DWDM limits to core networks and long-distance connections. [19-20]

3.7 TDM over Packet

TDM over Packet (TDMoP) describes technologies that are used to emulate circuit switched traffic like TDM E1s or STM-1s in packet network. The underlying packet network may be based on, for example, Ethernet, MPLS or IP technology. TDMoP is an especially important technology in network migration. With TDMoP virtual TDM connections are created through the packet network. These connections are called pseudowires (PW). Figure 15 describes a basic setup of TDMoP. Customer Edge (CE) devices CE1 and CE2 are TDM network elements. The Provider Edge (PE) devices PE1 and PE2 are converters that convert circuit-switched traffic to packet-switched traffic.

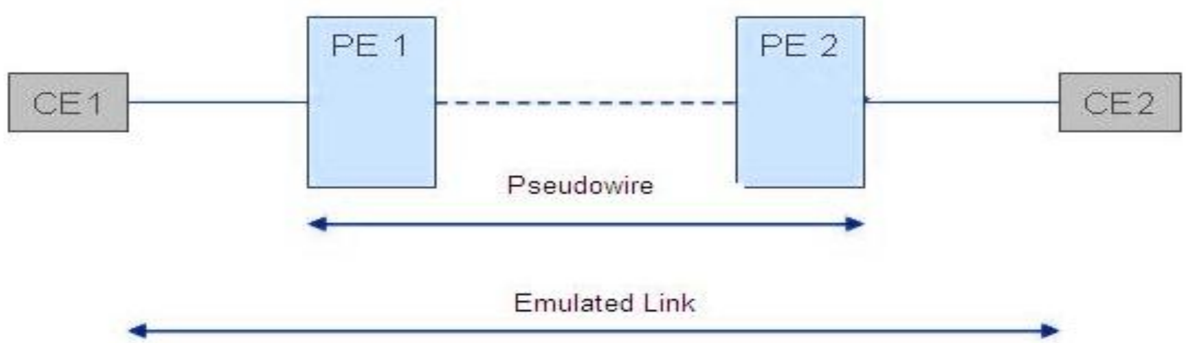

Figure 15 TDMoP

The simplest way to implement TDMoP is to encapsulate a E1 frame in a IP-packet by adding appropriate header to the frame. TCP/IP would provide a reliable way for encapsulation but it is not useful on voice transmissions because TCP resends packets that didn't reach the destination. This can lead to a situation where the voice packets could arrive out of order. That is why Real-time Transport Protocol (RTP) is more preferable in voice transmission. Encapsulation adds a lot of overhead to the TDM traffic but that can be handled with header compression and with the grouping of frames.

One of the main problems in TDMoP is how to obtain acceptable synchronization between PW endpoints. A good solution is to use a separate TDM-based synchronization network. Other options are to use GPS or to calculate synchronization information from delays between network nodes.

TDMoP related standardization is done in multiple organizations but mainly in ITU-T and in IETF (Internet Engineering Task Force). The IETF has set up a specific Pseudowire Emulation Edge to Edge (PWE3) group for developing the architecture for service provider edge-to-edge PWs and gathering information about different encapsulation techniques. A couple of common TDMoP technologies implemented in networks are:

- Structure-agnostic transport of TDM over Packet (SAToP).
- Structure-aware Time Division Multiplexed (TDM) Circuit Emulation Service over Packet Switched Network (CESoPSN).

SAToP protocol is used in multiplexing TDM streams like STM-1 over packet network. The protocol disregards structures imposed on streams like the standard TDM framing. SAToP is an ideal solution for networks where the packet network doesn't need to interpret the TDM data or to participate in TDM signalling. It is often used to transmit 2G data from mobile base stations to the network. CESoPSN is a quite similar protocol to SAToP. It is used to transmit structured TDM data over packet network. It also improves the resilience of the circuit-switched part of the network to effects of loss of packets occurring in the packet-switched network. With CESoPSN it is also possible to separate for example 64 kbit/s channels from the E1 frame. [21-22]

4. Migrated Services and Their Replacements

The users of the network services do not need to know what the underlying technology, in which the service is built, is. Different networks and technologies make it possible to provide different services. This chapter describes the most common services offered in PSTN and in NGN. The information about different services is useful in migration management, because it helps in determining suitable replacing solutions.

4.1 TDM Network Services

The TDM network services focused mainly on voice transmission. When the technology evolved, many additional features were implemented in these services. Technology evolution also made it possible to develop different usage purposes for the traditional telephone subscription.

4.1.1 Plain Old Telephone Service

Traditional telephone service is based on a bi-directional audio channel with a frequency range between 300–3400Hz. This allows telephone calls where both participants can speak simultaneously. More commonly this service is called Plain Old Telephone Service (POTS). Other basic POTS attributes are: call progressing tones, like a dial tone, and a ringing signal and emergency number service (for example 112 in Finland). The arrival of electronic switches in PSTN enabled including many additional services to POTS. These features can be divided into two categories: Customer Calling Features and Custom Local Area Signalling Services (CLASS).

For example the following services can be categorized as Customer Calling Features:

- Call waiting
- Call forwarding
- Conference Calling

Call waiting is used to notify users, who are already engaged in a call, that they are receiving an incoming call. Call forwarding, like its name implies, is used to forward calls to a different destination. Conference calling, also known as three-way calling, enables multiple persons to participate in a telephone conversation.

Custom calling features work basically on every phone. CLASS features on the other hand require SS7 features in order to work. A few of the more popular CLASS Display Features are listed below [23]:

- Caller ID
- Call Blocking
- Call Return

Caller ID enables the calling party's number to be displayed at the receiving end. This requires a device that is able to read the out-of-band signalling information that contains the number. Call blocking allows users to specify certain numbers from which he doesn't want to receive calls. These callers receive a message that their call is not accepted, while the receiving end doesn't get any indication of the call. Call return is used to return a call to the most recent caller. This returning call can be queued if the original caller is currently busy.

ISDN offers a similar service to POTS, but with some advantages. Many of the calling features were at first only available to ISDN based telephones, but today, when the technology has evolved, they are also possible to be implemented in PSTN. The existence of two B-channels offers the possibility to perform simultaneous functions. The user can, for example, use one 64kbps channel for Internet connection and at the same time another 64kbps channel for speech. Overall the digital technology is considered to offer higher reliability and better sound quality.

4.1.2 Other uses for POTS

PSTN and ISDN telephone services can be used in many purposes other than speech. One of the more traditional uses is to utilize POTS to access Internet. A modem is used to convert IP packets into audio frequency signals. Dial-up requires no additional hardware for the telephone network to provide this service, thus making it the most widely available form of Internet access. The downside of the dial-up Internet is the low transfer speed: the typical maximum transfer speed of most modern modems is 56kbit/s, which is a much lower rate than broadband Internet can offer.

Telephone subscription can be also used as a fax service. This can be done by connecting a telephone number to a printer or fax machine.

Other possible usage purposes of a telephone subscription are listed below:

- Alerting service (for example fire and burglar alarm)
- Payment terminal
- Traffic cameras
- Elevator phones
- Milking robots

As can be seen from the examples above, the range of potential solutions is quite wide. The purpose for which the telephone subscription is implemented can be an important factor when determining a replacing solution for service migration.

4.2 NGN Services

The convergence of different networks in NGN also enables the convergence of different services. Triple play is the term used in telecommunications market for describing the combined offering of three services: Television, Telephony (IP-based) and Internet. NGN enables the offering of triple-play services over the same broadband connection. It is estimated that currently over 80% of the revenue of incumbent operators is obtained through traditional voice [24]. The rise of IP-telephony has increased the vulnerability of operators, and that has encouraged them to offer bundled services including IP-telephony. Some operators have also planned offering a quadruple-play bundle that would include mobile voice and data in addition to normal triple play services. [25]

4.2.1 Voice over IP

Voice over IP (VoIP) is a technique for sending real-time, full-duplex voice over the Internet or intranet. It is a digital packet based technique. In VoIP the analogue voice signal is digitalized, encoded and then segmented into frames that are then stored into voice packets. These packets are then sent to the network and on their way to the destination they can travel through multiple switches and routers.

VoIP has several different advantages compared to the standard telephone services. Most of these advantages arise from the fact that VoIP operates on top of a packet switched network while the traditional POTS is deployed in the circuit switched PSTN. As discussed earlier, the packet network allows better utilization of bandwidth, because it is in use only when something is transmitted. Therefore more calls can be carried over a single link. VoIP also creates cost savings that are obtained mostly from the better bandwidth utilization. Another factor for creating cost savings is that VoIP requires fewer long-distance trunks between switches. That enables the billing to be based on the transmitted data instead of the distance used in the traditional service. VoIP also offers similar calling features as in POTS, like caller ID and call forwarding. These can be implemented at minimal extra cost. It is also possible to use an IP phone to call and receive calls from the PSTN. This can be achieved with adapters that translate IP addresses to phone numbers and vice versa.

The disadvantages of VoIP include packet loss and delay. Packets arrive to routers from many different sources and they are all queued for transmission over an outgoing link in the router. When the queue is full, the arriving packet is lost in the router, because there is no place left for it. When a lot of people are using the Internet at the same time, routers can become congested so that packet loss occurs. Packet losses can severely damage the quality of the voice signal. Several approaches for dealing with this problem have been presented [26]:

- Upgrading the network
- Silence Substitution – Substitute silence in the place of a missing packet(s)
- Noise Substitution – Substitute white background noise in the place of a lost packet(s)
- Repetition of Packets – The last correctly received packet is replayed in the place of a lost packet
- Interpolation of Packets
- Frame Interleaving
- Forward Error Correction – Packets are redundantly transmitted, so that a lost packet can be reconstructed from the subsequent packet

Transmitting voice in a packet network has some differences compared to data transmission. Data is considered to be delay tolerant but loss sensitive, while voice tolerates loss but is delay sensitive. That is why UDP is used to transport voice packets instead of the more traditional

TCP. There are many different sources contributing to the overall delay of the VoIP transmission. Couple of examples are listed below:

- Queuing delay, which occurs in different switches and routers where voice packets are queued behind each other to be transmitted over the same outgoing link
- Propagation delay occurs in a link and is the time signals require travelling from one point in space to another.

To summarize the VoIP, it can be said to offer the efficiency of packet-switched networks and at the same time it rivals the voice quality of circuit-switched networks. It creates cost savings for users and operators. One of the main reasons why VoIP hasn't been popular with telecommunication operators is the operator's need to maintain a healthy revenue flow obtained from the circuit-switched voice traffic. However, the rapid decrease of the PSTN revenues has increased the offering of VoIP. There are also companies, like Skype, who offer free VoIP calls between users. Operators need to address this situation by developing additional features and reliability to VoIP that customers would be willing to pay for. [27]

4.2.1 Session Initiation Protocol

NGN services use many different application layer protocols. Sessions Initiation Protocol (SIP) is one of the most important protocols and it is used in many NGN services. That is why a short introduction to SIP is provided.

SIP is a signalling protocol used in creating, modifying and terminating multimedia sessions where data is exchanged between participants. It is also used to invite participants to already existing sessions. Good examples of a session are a VoIP call or a video conference over IP. SIP itself doesn't carry any media data but it allows media to be added or removed from the existing session. It is an application level protocol that runs on any transport protocol. Like HTTP, SIP also uses text-based messages. These messages are divided in two categories: request from a client to a server, or a response from a server to a client.

The five different functionalities provided by SIP are listed below [28]:

1. User Location: Determines the end system that is going to be used in a communication.
2. User Availability: Determines the willingness of the called party to engage in communications.
3. User Capabilities: Determination of media parameters to be used.
4. Session Setup: Establishment of session parameters at both the called and the calling party.
5. Session Management: Transfer and termination of sessions, modifying session parameters and invoking services.

4.2.3 IPTV

IPTV is considered to be one of the main drivers of NGN deployment. It is a new potential source of revenue for telecom operators. IPTV in itself is not a replacement solution to any particular NGN service, but because of its importance a short overview of its characteristics is provided.

IPTV is primarily implemented by coding picture frames to IP packets and then multicasting them in the network. At the transport layer IPTV uses UDP instead of TCP. The reason for this is the same as with VoIP: to reduce delays. IPTV requires a broadband connection in order to work properly. For example a High Definition Television (HDTV) approximately requires a 20 Mbps connection per channel and in times of channel changes about 40 Mbps/channel is required for two channel streams. In the last few years the broadband Internet has become more widely available and the number of IPTV subscribers is growing steadily as well. In 2009, estimation was that by the year 2013 there would be 115.6 million IPTV subscribers [29].

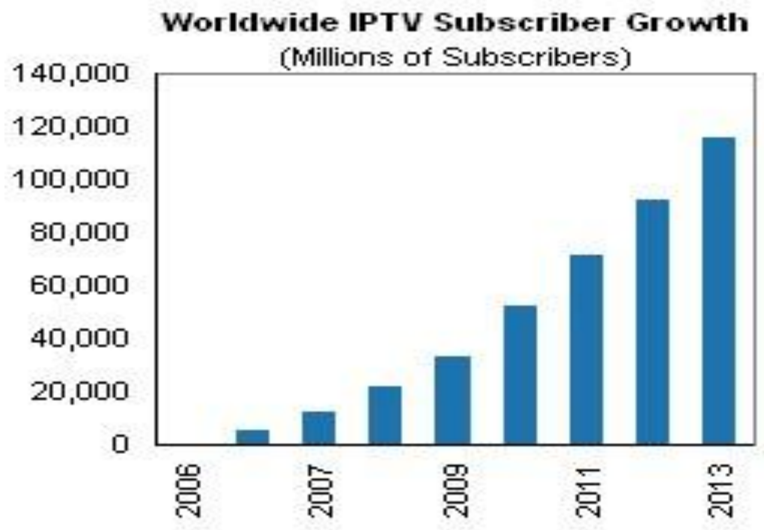

Figure 16 IPTV subscribers

Based on the data dated 2010, Europe has the largest IPTV subscriber base, but in other countries, especially in China and the USA, the subscription amounts have grown rapidly. In China the subscriptions have doubled during just one year [30].

Table 3 IPTV subscriptions top 10 countries

Country	**2009Q1**	**2010Q1**
France	7 066 000	9 018 305
USA	4 171 850	6 071 898
China	2 850 000	5 750 000
South Korea	1 450 407	2 576 663
Japan	1 340 608	1 861 127
Germany	740 000	1 522 500
Hong Kong	1 140 000	1 165 000
Russia	700 000	1 117 900
Italy	790 000	825 000
Spain	711 390	825 000

IPTV has many advanced services and features compared to the traditional cable or terrestrial TV. One interesting feature is time shifting, that allows users to replay, pause, and rewind TV broadcasts. Other important feature is the Video on Demand (VOD) service, which allows users to browse and access different videos on the Internet.

4.2.4 Other NGN services

In addition to VoIP and IPTV, NGN provides multiple different services. The actual service portfolio depends on the service provider, and this can vary in different countries or areas.

One good example of an NGN service is the CStream service provided by TeliaSonera in Finland. CStream offers a two-way messaging channel for different information. The messaging channels supported are listed below:

- SMS
- Fax
- Voice Message
- E-mail

To transmit information with this service, the user sends a text file containing the message they want to send and an identification number of the receiving end (for example a phone number, a fax number or an e-mail address). CStream then modifies this text file to the desired format (SMS, fax page or voice message with text to speech technique) and sends it to the receiver. The service has a web-based application where the received messages can be viewed.

CStream is a useful service in migration, because it can be used to replace traditional fax services. It also enables businesses to manage their different messaging channels from the same portal.

Before the implementation of NGN, the businesses handled their inner communications with telephone switches. Typical products that operators offered to customers were 2 Mbps E1-lines with ISDN PRI interface. This central line was then used to handle the company's telephone calls. TeliaSonera offers a service called Sonera Office Communications to replace 2 Mbps E1s used by businesses. Sonera Office Communications offers a common platform for all real-time messaging like voice calls, meetings, content sharing and instant messaging. VoIP is used to provide voice calls in this service. There are also additional features, like information about users present in the network. The Sonera Office Communications service can be used with computers, laptops or with certain mobile devices.

These were just a few examples of different services enabled by the emergence of NGN. There is a constant process going on to create new services and further improve the existing products.

4.3 Wireless replacement solutions

This thesis mainly focuses on how the TDM network services are migrated into the fixed IP-network. There are also situations in which a sufficient IP-product is not available, for example when the whole wire network is replaced. In some cases it is more cost efficient and practical to replace a TDM service with a wireless solution. That is why a short introduction of different wireless services, used in replacing TDM services, is provided.

Probably the simplest example is to provide a traditional mobile phone subscription to replace the POTS connection. Mobile phones have acceptable voice quality and provide additional mobility compared to the POTS. The downside of this solution is that customers need to change their telephone numbers because of the different numbering schemes between fixed- and mobile networks.

TeliaSonera has implemented a service called Home Number, where the connection is based on a mobile technology, but customers can still have their old telephone number. Basically every Home Number subscription really has a mobile number but a network server converts it to a traditional telephone number. Customers can use this subscription with a mobile phone or with a phone that resembles a traditional landline phone. The drawback of this solution is that it isn't suitable for all of the services the POTS were able to provide. For example a fax service or a payment terminal can't be implemented with Home Number.

TeliaSonera has also developed a wireless replacement for telephone switches. Sonera Mobile Centrex service offers a similar service to the traditional TDM switches. The only difference is that it is implemented in a wireless environment. Between 5 and 200 mobile subscriptions can be connected to the Mobile Centrex.

5. *Migration* to NGN

This chapter examines different reasons that why network migration must be done. The reasons range from the decreased usage of traditional telephone services to increased usage of data services. Migration can be done in different ways and these rely heavily on each other. These different implementations are also described in this chapter.

5.1 Reasons for Network Migration

The network examined in this thesis was originally built to accommodate more than one million telephone customers. During the last two decades, the mobile technology has evolved rapidly, and that has led to a decrease in the usage of traditional telephone networks. Because of that, the utilization levels of TDM equipment, like concentrators and switches, have also decreased. That has led to decreasing profits.

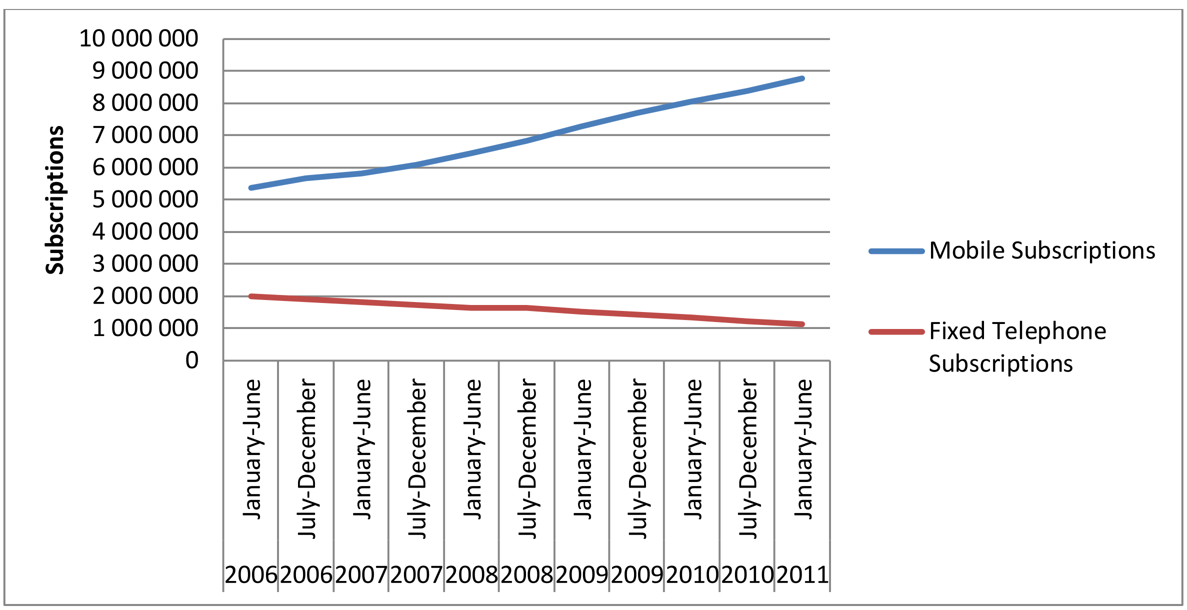

Figure 17 The amount of Fixed- and mobile telephone subscriptions [31-32]

Natural churn away from the traditional voice services is continuing, so today the utilization rate of some concentrators is as low as 1%. It is estimated that with natural churn the migration from TDM to IP would be completed in approximately 2030. It is easy to understand that maintaining a network part with a utilization rate of 1% isn't economically viable. That is why the telecom operators must participate actively in order to speed up the network migration.

Finland is a very sparsely populated country. Population density is only 16 persons per km^2, which is the third-lowest population density of any European country. The population is also distributed very unevenly, with the majority of population concentrated on the south-western coastal areas [33]. Because of this uneven distribution, it isn't always economically sensible to maintain fixed copper lines in the more sparsely populated regions. The data has to travel long distances along the wires, which leads to an increased signal attenuation and decreasing data rates. In these more rural areas wireless networks can be deployed to offer a service similar to copper lines. The GSM coverage in Finland is close to 100%, so the traditional telephone services can be replaced by similar mobile services almost everywhere. The third generation (3G) mobile network, which can offer data rates of up to 6 Mbps, is constantly expanded and it can be used to replace ADSL-connections. There is also a possibility to replace fixed-line wires used in ADSL-connections with a satellite based service.

Removal of the fixed network in rural regions also creates different cost benefits. There are hundreds of thousands of poles in Finland to which these copper wires are attached. These air-cables are very vulnerable against different environmental conditions, like heavy snow or falling trees. Repairing of the damages caused by the bad weather can be quite difficult and expensive, because the repairing teams have to travel long distances and the wires go through difficult terrain. The poles also suffer and rot during the years. On average, the poles must be changed every 40 years, and replacing all the poles would be a massive investment. Today, when the wireless networks offer practically equal service levels compared to the wired networks, it is no longer profitable to invest in the new poles replacing the old ones.

In more densely populated regions where the majority of cables have been dug into the ground, it is more practical to replace PDH and SDH connections with switching and routing equipment enabled to handle the packet traffic. Traditionally the major part of the traffic in the networks has been voice. The voice traffic has also created the majority of profits for network operators. In the 1990's when the Internet became more popular, the amount of data traffic in the network started to increase exponentially and this trend is still continuing today. This is happening around the world. For example, Figure 18 shows how the amount of network traffic has grown in North America and how it is predicted to grow in the next few years.

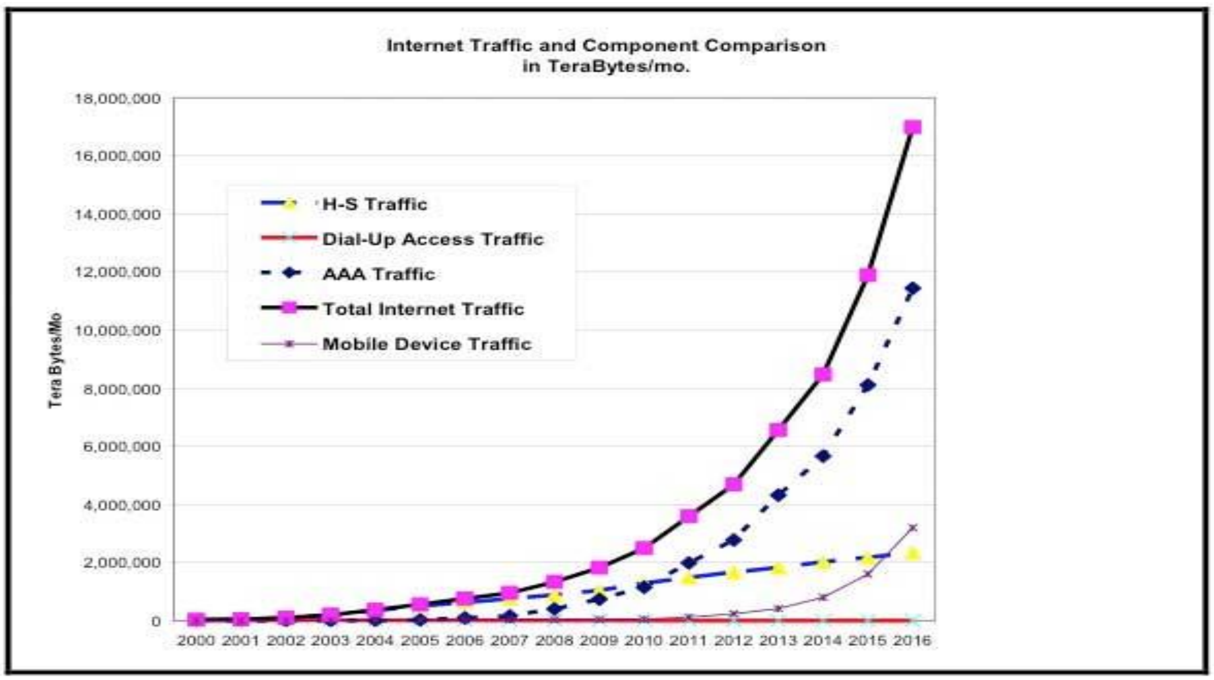

Figure 18 Change in traffic volumes [34]

In Figure 18 it can be seen that the traffic from the different fibre-based advanced access architecture (AAA) lines is the main source of traffic increase. The yellow-blue line indicates growth of the High-speed (H-S) access lines like xDSL and Cable Modems. The growth in there is much more moderate compared to the AAA traffic. Figure 18 indicates that Internet traffic from mobile devices surpasses the traffic amount of the H-S lines in 2015. The Dial-Up Access Traffic continues to remain close to zero. There have been many other studies that have predicted similar increase in the Internet traffic. For example a recent Cisco white paper [35] indicated that the global IP-traffic has increased eightfold between the years 2005 and 2010 and the prediction is that the annual growth rate of IP traffic from 2010 to 2015 is going to be 32%.

TDM-based services like PDH and SDH were originally designed to transmit voice. SDH especially has been improved to also transfer data, but still the Ethernet based solutions offer lower costs and better performance. This is why network operators have built a separate network for data in parallel with the traditional TDM network. Both networks create discrete maintenance and monitoring costs. NGN provides technological improvements that allow convergence of different networks, so that voice, data and video can all be transferred on the same platform. Combining of the separate networks reduces the capital and operative expenses that were related to the maintaining and managing of the different networks. NGN makes it possible to deploy new more advanced services like video conferencing and Virtual Private Networks (VPNs), which allow organizations to combine their existing private networks with the portions of PSTN. These new services help network operators to create more profit to compensate for the decrease of profits in the traditional voice. They can also be implemented quite rapidly, which reduces the time to market and the life-cycle costs of offering new services [36].

The different network element providers have also recognized the changing environment. Most of the manufacturers are planning to or have already stopped the manufacturing of TDM network elements. This has led to a situation where it is hard to find enough spare parts to maintain the TDM network. The vendor specific support for the TDM devices also isn’t always available anymore. In Finland there is a situation where the majority of the current network engineers are approaching retirement in the next few years. Today the education focuses on packet-based networks, so TDM knowledge is slowly beginning to disappear.

5.2 Different ways to migrate products

5.2.1 Natural Migration

In this context natural migration is used to describe a situation where customers independently terminate their fixed-line telephone subscriptions. In an ideal situation they would also subscribe to a new IP-based service, even though today people generally only have a mobile telephone subscription. The good thing about natural migration is that it doesn't require any investments from the network operator. Operators can still encourage natural migration with different marketing campaigns and price reductions. The problem with natural migration is that it is difficult to control and predict. It also happens randomly around the network, so that in the end multiple network areas could exist, where only a few customers remain. A spreadsheet tool presented in Chapter 7 was developed to help with predicting natural migration. When considering the economic aspects and availability of spare parts, natural migration is happening too slowly.

5.2.2 Active Migration

Targets of active migration are to maintain a high customer satisfaction level and to migrate their TDM based solutions to IP or wireless solutions. In active customer migration each customer within a target area is contacted and notified about the network migration. Contact persons assigned to the migration project then interact with a customer to find the best possible solution to migrate the customer's connections. When doing an active migration a detailed analysis about the customers and their subscriptions are needed. These analyses and customer contacts generate additional expenses and create the need of personnel dedicated to the project. That is why active migration focuses on network areas with a small amount of customers. It is preferable to maintain the yearly active migration rate at the same level, so that the same experienced personnel can handle the contacting-replacement-dismantling process during the whole project. If the active migration levels suddenly increase, new employees need to be hired and trained. This can cause problems if the migration amounts return back to the previous level. The active customer migration levels can be controlled with emulation.

5.2.3 Emulation and packing

A technique where TDM-traffic is transmitted through the IP-network is called network emulation. In emulation the end points of the data traffic use TDM, while in the middle of the connection the data stream is converted to packet format. This way services that rely on TDM or services that do not yet have a replacement solution can be migrated. Emulation also frees a lot of different PDH and SDH transmission systems, because only the TDM devices at the end points must remain in operation. Emulation can be done in different ways. One common solution is to use the TDMoP technology. Using the TDMoP requires investments in the emulation devices. Some TDM network devices can be connected straight to the IP core network. However, this requires a bit of configuration on the network side, and that creates additional costs.

Because of the additional device investments and configuration costs, emulation is usually implemented in network areas that have more customers. These areas still usually create revenue, so it is economically sensible to keep maintaining the TDM-network. Even in these denser parts of the network the natural churn is decreasing the subscription rates. This development should be carefully monitored, because some emulated areas can be transformed to potential candidates for active migration. This then frees emulation devices that can be deployed in some other part of the network.

The original TDM network in Finland was built to accommodate more than one million telephone subscriptions. Today there is a significantly smaller amount of active subscriptions and this has led to a situation where there is redundant capacity on some network sites.

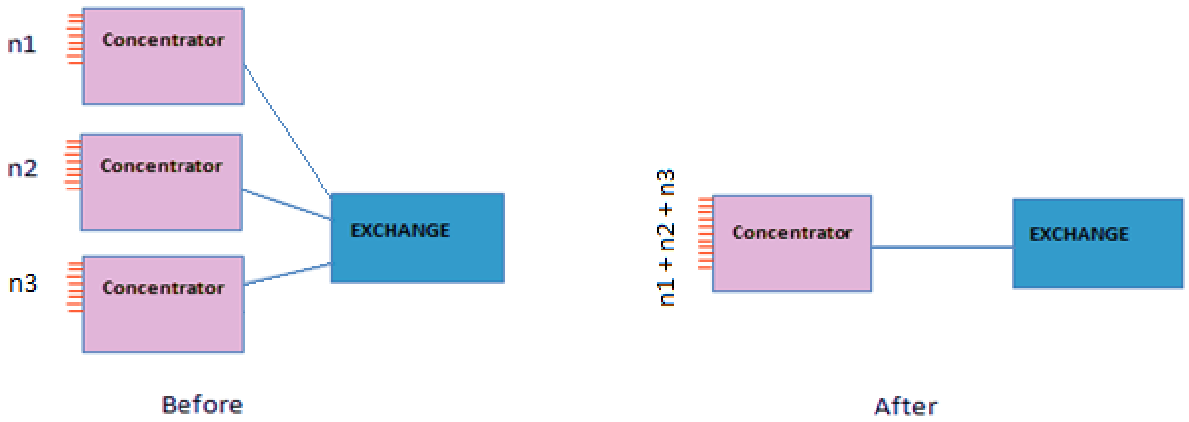

Figure 19 Concentrator Packing

For example, on a site there can be three telephone concentrators each of which can handle 500 connections but each only has 150 active connections. In this situation, concentrators can be packed so that all 450 connections go through a single concentrator. That makes the other two concentrators redundant, so they can be dismantled to free up resources.

6. Migration analyses

Migration analyses are a key factor in migration projects. Analyses are used to optimize the migration process to create cost savings and maintain customer satisfaction. This chapter describes how these kinds of analyses can be formed. Another point that greatly affects the migration project is the migration strategy. This chapter shows a few different strategy scenarios and analyses the pros and cons of each approach. Finally an example of a practical migration problem is presented.

6.1 Overview of a Migration Planning Process

An ideal migration project should be performed so that it would cause a minimum amount of disturbance to the customers. For this purpose natural migration is the ideal way, but from the migration's point of view it is going on too slowly. That is why thorough analysis and monitoring processes are needed so that the migration is implemented as smoothly and efficiently as possible.

In migration analyses there are different points of view that need to be taken into account. The network side of the organization usually has its own preferences about the dismantling order and timetables, and these can often conflict with those of the business side. The network operator's main motivation for the migration order is to maintain sufficient storages of spare parts and make sure that the TDM network remains operational during the transition process. At the same time the business side wants to uphold the customer satisfaction, especially with the key customers that bring in a lot of revenue. The product management also has its own opinions about the migration schedule. It mainly depends on the release schedules of the replacing products. Different analyses are needed in order to reach a solution that satisfies all participants. An example of an analysis that takes different needs into account is presented in Chapter 8.

In Figure 20 the key parts of the migration planning process are described. The first part of the process is to obtain the necessary data from the different databases. This can be roughly divided in the network data that describes the network topology and gives information about different network devices, and the customer data where information about the customer types and products are found.

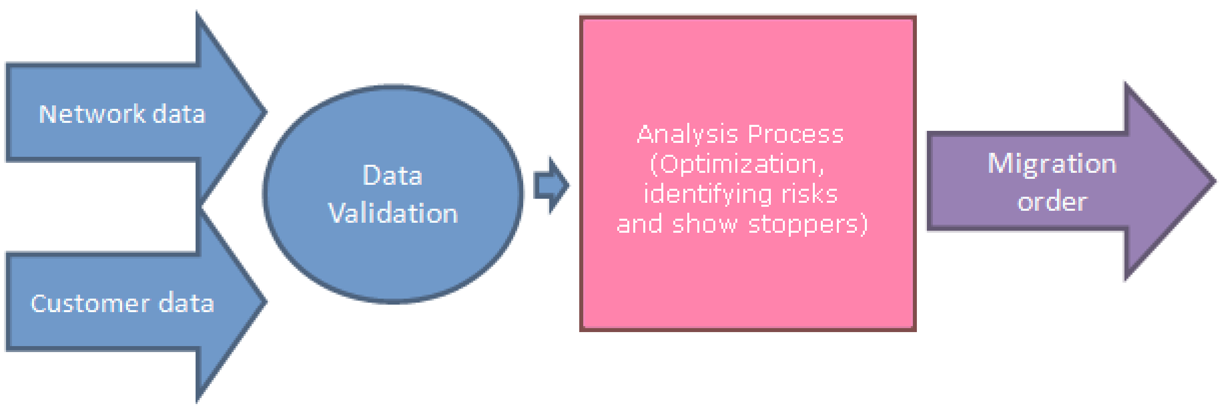

Figure 20 Migration Planning Process

The data is obtained from different databases that are built on different platforms and are not interconnected. That is why the data can be in different forms and qualities depending on where it is located and who has modified it. For example, in network data every connection has a starting point and an ending point. Let us assume that the network operator’s policy is to always consider a concentrator to be the starting point of a connection. This is a general working model, but still there are some network configurators that during the years have set the concentrator to the ending point of a connection. This then lead to a situation where these connections are hard to identify, especially when analysing tens of thousands of connections.

That is why the data amount needs to be validated and modified, so that it is in an easily interpretable and high quality format, before it can be used in analyses. Data validation is a major cost contributor in the network migration. The amount of corrupted data needs to be identified and its effect on the migration examined. During the planning process, the migration management needs to decide about the scope of data validation:

- If no data validation is done beforehand, there is no upfront cost, but the costs during the migration would probably be higher because the decisions made could be based on corrupted data.
- If complete validation is done, all costs are upfront, which could become too expensive, but at the same time the migration costs would be much lower. [37]

The analysis process is used to determine the migration order, i.e. which products and subscriptions are migrated and when. When the migration is started there isn't necessarily a replacement product available for some old platform services. Definition of a replacement product is the key to successful migration for each customer. Examples of a situation into which product migration can enter are:

- One IP-service is used to replace three TDM-services, or alternatively an existing IP-service can only partly replace some TDM-services. These cases must be identified and if there isn't a satisfactory solution available, the dismantling order must be changed.

- A customer can use their subscription to whatever purpose they want. For example the traditional telephone subscription can be deployed in traffic cameras and milking robots used on farms. This information can't always be easily found in the network provider's databases, so customer interaction is needed. The availability of a replacing solution could depend on what purpose the product is used for. Therefore this needs to be identified in the analysis process.

The information about the customers is also an essential factor in the analyses. Generally the business customers create more revenue compared to the consumer customers. Business customers can also have multiple, sometimes hundreds of subscriptions around the country. These cases require more time and effort compared to regular consumer customer subscriptions. That is why these cases need to be identified early, so there is enough time to perform the necessary actions. Some examples of the customer factors that need to be considered in order to validate the migration opportunities are listed below:

- Whole value of the customer (revenue)
- Customer Solution and can it be replaced
- Customer contract
- The amount of customers in a network area

These factors are the basic building blocks in determining the optimal migration order. A closer look into this process is provided in Chapter 8.

6.2 Choosing the right migration Strategy

The migration from PSTN to NGN is a long and complicated process. This section examines different migration scenarios. The important thing is to understand that there is no single best way for the migration to NGN, because of the different underlying conditions. For example, the most suitable migration scenario could depend on the current network infrastructure and service utilization rate. Arvind Kumar proposes three different types of migration scenarios in his paper [38]. This model is presented in Figure 21. Other migration scenario models have been also proposed [39-40].

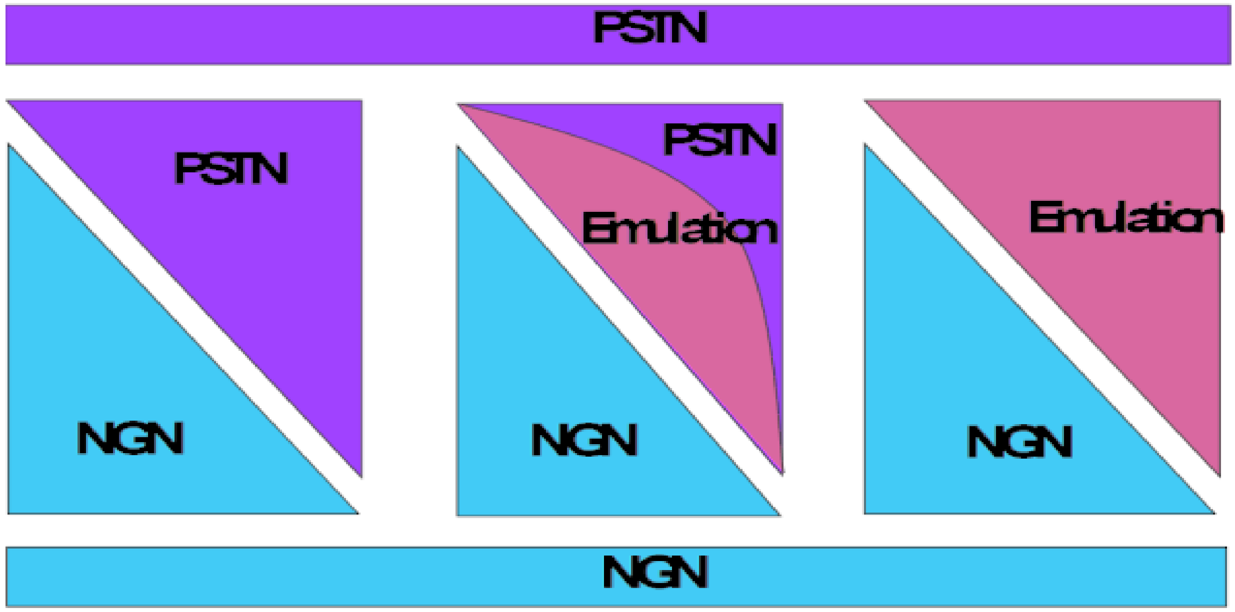

Figure 21 Different Migration Strategies [Adopted from 37]

The first strategy describes an overlay migration scenario. In this scenario NGN and PSTN are operated in parallel. The NGN usage is slowly increased as customers migrate away from PSTN. Finally, when there are only a small amount of customers left in PSTN they are migrated to NGN. The overlay strategy's advantage is that it allows a rapid implementation of new services and minimizes the risk of disrupting the existing PSTN customers. The major disadvantage of this strategy is that it doesn't reduce network costs because PSTN network needs to be maintained and building a separate IP network is necessary.

The scenario on the right in Figure 21 describes a PSTN replacement scenario. In this scenario a PSTN infrastructure is replaced with IP based equipment. During the transition period the PSTN services are enabled in NGN through emulation. Just like in the overlay scenario, the NGN usage increases as customers naturally migrate away from PSTN. At the same time emulated connections can be dismantled. This scenario is usually preferred by incumbent operators who have a large PSTN infrastructure. The biggest obstacle of this approach is successfully managing all of the old platform connections and making sure they remain operational during the transition period.

The third solution is a mixed scenario, where the network infrastructure is partially replaced. Some of the PSTN connections are emulated while some are actively migrated to NGN. This is also the strategy selected for the migration project presented in this thesis. The main idea of this strategy is that profitable and otherwise critical PSTN connections are emulated while the others are actively migrated. At the same time the natural migration increases the use of NGN services. The main advantage of this approach is that it is optimal in obtaining maximum customer satisfaction. New services are provided according to a customer's interest in them, while at the same time support for the old platform services is provided until a suitable replacement is developed. This scenario also allows taking the maximum advantage of old TDM-devices by reusing them in different sites as the migration progresses. The major disadvantage is that this approach requires a lot of management and optimization, which can create additional risks.

6.3 The Practical Migration Example

During the first years of the migration project the easiest targets are identified and migrated. A typical easy case is a telephone concentrator where there are approximately 10 POTS subscriptions and a few or no ISDN subscriptions. From the transmission technology point of view, the ISDN subscriptions don't differ from the POTS, but they can be considered more complicated because they usually belong to businesses. Emulation isn't a cost-effective solution because there are so few customers. The network view of the case is presented in Figure 22.

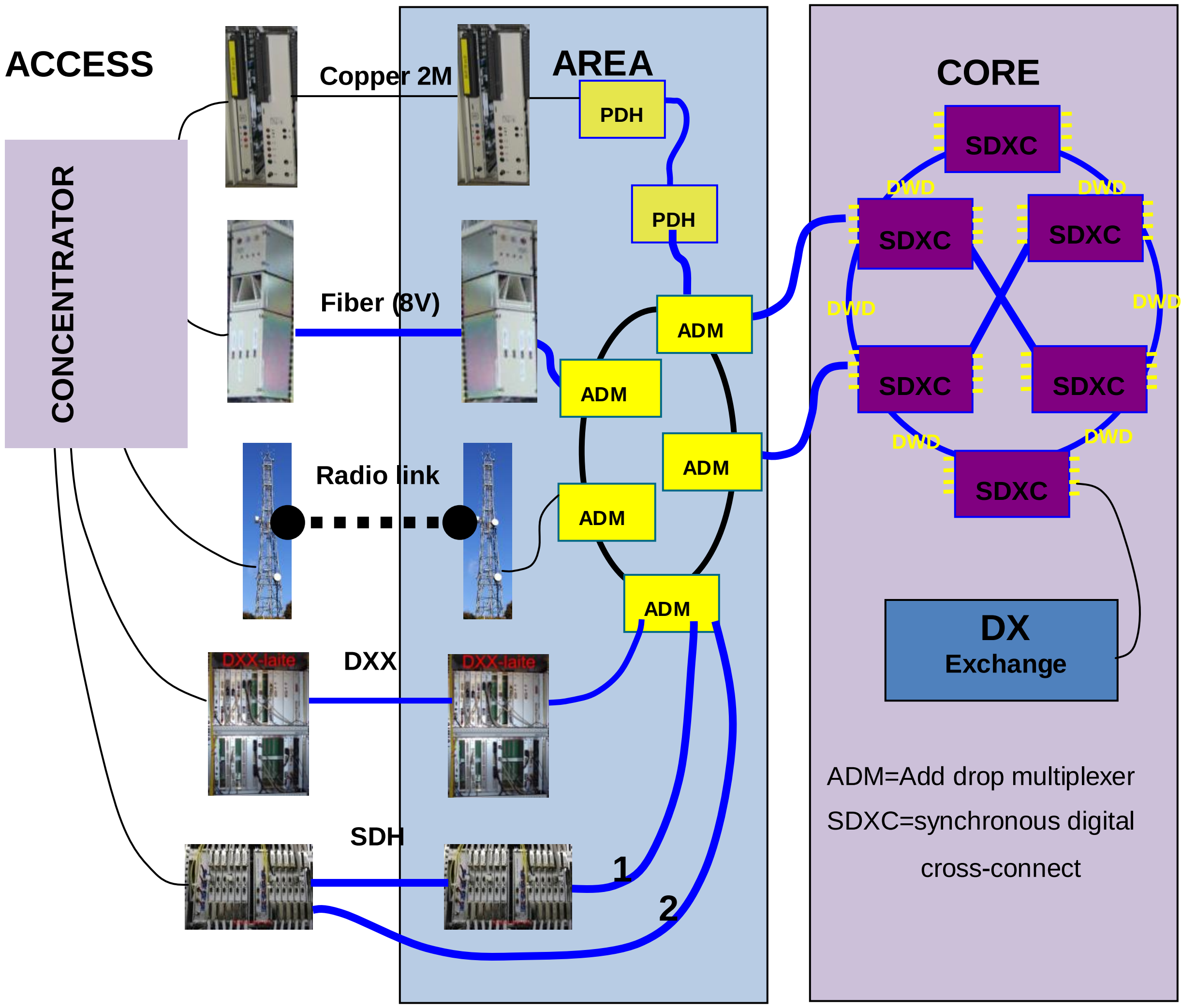

Figure 22 Network view

Figure 22 describes different ways how a concentrator can be connected to the area and core networks. When the concentrator is emptied of the customers, the transmission technology connecting the concentrator to the area network can be dismantled. These individual cases do not affect the area and core networks. In more complicated situations there are also 2 Mbps company connections on the network site. These are also connected to the area network with the same transmission technology as the concentrators, so the same emulation and product migration possibilities still apply.

A typical yearly migration target is a few hundred concentrators. That releases several thousands of transmission technology devices. The key point is that by emptying a concentrator, many other devices can also be dismantled. The decision of which concentrators will be emptied requires analyses concerning the existing subscriptions and customers.

7. Example Case 1: An Analysis Tool to Predict and Manage Future Migration Amounts

During migration planning it is important to be able to predict and manage different migration amounts. For that purpose a specific spreadsheet tool was created. This tool shows how fast different TDM based products are migrated and how it affects revenues and costs. The tool was built with Microsoft Excel, which was considered to offer the required attributes for this purpose. The underlying principles and formulas behind the tool were quite simple to implement, so any specific analysis program was not needed. This chapter describes the building process of the tool and how it is used in migration management.

7.1 General Description of the Tool and How It Was Built

The first step in building of the tool was to determine different products to be analysed. In this case the scope consisted of the majority of the TDM product portfolio. This includes basic telephone subscriptions for consumers and businesses, ISDNs, 2 Mbps switches of different technologies and wholesale products. To provide the necessary simplicity, each product is examined on a separate sheet. A specific summarization sheet was created, where the different key numbers are displayed for each product.

The next step was to obtain the current amounts of different product subscriptions. This information lies around different databases, but they were quite easily gathered. When the product amounts were obtained, different categories to represent where and how products are migrated were created. After several trial versions the status description of the product was divided in the following categories:

- The amount of subscriptions of the product
- The remaining TDM-base
- The emulation base
- Natural migration
- Active migration
- Yearly emulation

The tool follows and predicts how the amounts in these categories change during the lifetime of the migration project. The categories can be examined on a yearly, quarterly or weekly basis, depending on the migration task. The amount of subscriptions of the product simply tells how many subscriptions are left of the product examined. It is a sum of the TDM base and the emulation base. At the beginning of the product migration all of the examined subscriptions are part of the TDM base, which means that they are built on top of TDM technology. When the migration proceeds, some of the products are emulated to IP network, so that during the migration the TDM-base is decreasing, while at the same the emulation base is growing larger.

The first year's natural migration amount can be calculated, when the number of subscriptions there were the year before is known. That can then be compared to the current amount to obtain the decrement of subscriptions. This rate can be used to help predict future natural migration numbers. Later on in the project it is important to remember that there is also a churn away from the emulated base of the product. When the service is emulated, the customer will not be influenced, so it doesn't affect their potential decision to terminate the service. Total active migration and emulation rates are determined by migration management. These rely heavily on the network operator's knowledge about the network quality, capacity utilization and need for spare parts. As mentioned earlier, the active migration and emulation complement each other. The network operator simply indicates how many network elements it needs to dismantle, and then the migration management needs to decide how many of these elements can be obtained from emulation and how many from active migration. These numbers also depend on natural migration, so if the natural migration rate is higher than expected, the emulation and active migration rates can be lower. Figure 23 describes the simplified category overview of the tool. The amount of subscriptions, the TDM base and the emulated base indicate the amounts at the end of the year. The migration amount shows how many subscriptions are migrated during the year.

Table 4 Product x subscriptions

Product x subcriptions 31.12.2010: 200	2011	2012	2013	2014	2015
Product x subscriptions	**160**	**128**	**102**	**82**	**66**
TDM base	156	120	77	49	25
Emulation Base	4	8	25	33	40
Natural Migration Total	**40**	**22**	**16**	**10**	**6**
Natural Migration from the Emulated Base	**0**	**1**	**1**	**3**	**3**
Natural Migration from the TDM base	**40**	**21**	**15**	**8**	**4**
Yearly Emulation	**4**	**5**	**18**	**10**	**10**
Active Customer Migration	**0**	**10**	**10**	**10**	**10**

From Table 4 it can be seen that for product x the measurement period started at the end of the year 2010 when there were 200 subscriptions. During the year 2011 40 subscriptions churned away, while 4 were emulated. In 2012 the forecasted emulation is 5 subscriptions, but at the end of the year 2012 the size of the emulated base is only 8, because it has been predicted that one subscription from the emulated base churns away during the year. The goal for the migration project is that when the project ends, the size of the TDM-base would be 0. After that a new project plan and schedule need to be created for migrating the emulated base.

In addition to the information about how the subscriptions are migrated and how the migration is estimated to proceed, it is important to know what happens to the customers who have been actively or naturally migrated. Some customers may subscribe to a replacing IP-solution, some may move to a similar wireless solution and some may choose not to have any replacement product. In the building process of the tool, replacing solutions for every examined TDM product have been determined. After that the amounts of every replacing solution were obtained from databases and brought into the tool, so that the active and natural migrations' effects can be seen. This is especially useful after the first year of the project. Then the actual amounts of replacing solutions are known, so that their percentile distribution can be calculated and these percentages can be used to make forecasts for the following years. Table 5 describes this forecast process. It presents a replacing product distribution of the product x studied in Table 4.

Table 5 Distribution of replacement solution

	2011	2012	2013	2014	2015
Natural Migration Total	**40**	**22**	**16**	**10**	**6**
Distribution of replacement solutions					
IP solution 1	**5,0%**	**5,0%**	**5,0%**	**5,0%**	**5,0%**
IP solution 2	**42,5%**	**42,5%**	**42,5%**	**42,5%**	**42,5%**
Mobile solution	**22,5%**	**22,5%**	**22,5%**	**22,5%**	**22,5%**
No replacement	**30,0%**	**30,0%**	**30,0%**	**30,0%**	**30,0%**
Amount of replacement solutions					
IP solution 1	**2**	**1**	**1**	**1**	**0**
IP solution 2	**17**	**9**	**7**	**4**	**3**
Mobile solution	**9**	**5**	**4**	**2**	**1**
No replacement	**12**	**7**	**5**	**3**	**2**

At first the actual amounts of replacing solutions for the year 2011 are obtained in the beginning of 2012. After that their distributions are calculated and the same distribution is used to predict the future amounts. This forecast model can then be updated and re-evaluated every time new information is received. One flaw of the tool can be found when looking at the numbers of the year 2013. The forecasted natural migration is 16 subscriptions, but when looking at the replacing solutions, their sum is 17. This small defect arises from the fact that when talking about subscriptions, it is not sensible to add decimals to the numbers. Now when the total amount is multiplied to the percentage values, there will be decimals that Excel rounds up to the nearest number. That is why there is a possibility for this kind of errors, but it shouldn't cause any significant problems in migration planning.

7.2 Usage in Migration Management

Knowledge of the migration amounts and product distributions have many uses in the migration management. Perhaps the main advantage of the tool is that it allows the management to visualize, how migration is progressing for each product and what needs to be done in order to achieve the targeted goals. For example, when looking at the product x's current migration forecast it indicates that at the end of the year 2015 there are still products in a TDM base. Management can now then react to the situation in several possible ways:

- One solution is to encourage natural migration with price changes and marketing campaigns.
- The active migration and emulation rates can possibly also be increased. They often depend on other products, but this relation can be easily analysed with this tool, because the related products are presented on separate sheets.
- If none of the mentioned solutions are possible to implement, the final solution would be to increase the project lifetime. The formulas used in the tool allow a possibility to easily add additional columns in products.

One good usage of the tool is to use it to approximate yearly revenues. These approximations can be done, when the Average Revenue per User (ARPU) for the examined products are known. Different replacing solutions have different ARPUs and these also differ from the ARPU of the TDM product. Table 6 presents the ARPUs related to the product x. As can be seen, the tool offers a possibility to change ARPUs during the product lifetime, but to simplify things in this example, the product x's ARPUs remain the same the whole time during the project.

Table 6 ARPUs of examined products

ARPUs	2011	2012	2013	2014	2015
Product x	20	20	20	20	20
IP solution 1	10	10	10	10	10
IP solution 2	13	13	13	13	13
Mobile solution	15	15	15	15	15

With these numbers, the yearly revenue streams and their changes can be calculated.

This monitoring tool is also deployed in calculation of the costs related to active migration. The employee cost of active migration can be roughly divided in four categories:

- Customer contact
- Customer analysis
- Service disinstallation
- Equipment dismantling

In active migration each customer must be contacted and analysed to find optimal solution, which would benefit both parties most. When the solution is reached, the old services must be disinstalled and after that the related equipment can be dismantled. A separate analysis has been made of how much time approximately goes to each phase. Now when the average cost per employee is known, the cost of dismantling a subscription can be calculated.

From the previous examples it can be seen that this kind of monitoring tool has many benefits. The migration management needs to be aware of how migration is progressing and how it is forecasted to proceed in the future. This information helps to identify different problem points related to different products and gives additional time to manage them. One important feature of the tool is that it is very flexible. Probably in the future new applications will be developed, where the information provided by the tool is exploited

8. Example Case 2: Determining the Optimal Migration Targets

In the migration project, the main motivation comes from the network operators need to have enough spare parts in order to maintain the TDM-network, during the evolution towards NGN. A common case example is the dismantling of the telephone concentrators. Concentrators are devices used to gather the subscriber lines and connect them to a local exchange. The most typical migration case is a small network site where there are only PSTN customers. If the concentrator is dismantled on the site, the PDH and SDH transmission systems can also be dismantled. In this example it is examined how the yearly migrated concentrators are defined and what kind of analyses are needed in identifying the targets.

The analysis process requires a combined effort and co-operation between the network- and service operator. The planning process begins within the network operator. The network informs the migration management of how many concentrators it needs to be dismantled in the next year. The management must then use this as a starting point to define the potential targets. This prioritization is done by giving the concentrators a specific prioritization order based on different factors. Important thing is to understand that the described concentrator dismantling project doesn't focus on a certain region or area but the target area is the whole TDM network.

8.1 Network Operators Point of View

The network operator tries to determine in its analyses how many concentrators it needs to dismantle in a year. These concentrators can be obtained with packing and with active migration. The potential packing targets are all of the network sites that have more than one concentrator. These sites are then analysed to see how many connections there are connected to the concentrators. If the utilization rates of the concentrators are low enough they can be packed. After the packing amounts are known, the rest of the target devices are emptied with active migration.

The list of how many connections are connected to a concentrator is obtained from the network operators databases. They have also some preferences about what devices should be emptied. These preferences rely heavily on what is happening in the other parts of the network. There are for example large telephone exchanges where multiple concentrators are connected. When these kind of larger switches are replaced with an IP-solution, necessary actions must be done with the concentrators. Some of the IP based switches like the Surpass hiE9200 [41] also supports TDM switching but connecting concentrators to it still have costs. Other solutions are emulation with the TDMoP, especially with larger concentrators, and active migration with smaller concentrators. There are also concentrators from different vendors in the network and not all will support emulation. These must be prioritised in active migration.

The network operator wants also to analyse what kind of PDH and SDH devices can be dismantled when the concentrator is emptied. Concentrators are connected to the area network usually with PDH or SDH devices and these can be also dismantled when the concentrator is empty. Some devices have less spare parts or vendor support available so these cases must be taken into account when the network operator defines its preferred targets.

To summarize the network operator's point of view, it wants to prioritize the concentrators connected to the exchanges, which are transformed to IP in the near future. Also the concentrators that do not support emulation receive higher prioritizing order. In a typical yearly scenario the network has defined that 300 concentrators need to be dismantled. 150 of these can be obtained with packing so the rest are targeted to active migration. The network then sends information about its preferred targets to the service operator for further analyses. Even though the target amount for the service operator is 150 concentrators the prioritization is done for larger amount, for example for 600 concentrators. This is done so that there would be a buffer for the potential risk and show stoppers that could arise from the service operator's analyses.

8.2 Service Operations Point of View and Identifying the Targets

The starting point for the service operator's analyses is to identify what customers and products there are behind the 600 concentrator list obtained from the network side. The customers are divided in the consumers and the businesses. The businesses are then classified into three categories based on the contract values and the total amount of subscriptions. The level 1 customer are considered to be the most valuable while the level 3 includes business customers of lowest importance. After the classification is completed, a specific analysis for the level 1 customers is done. This analysis examines how many subscriptions businesses have in the target area of 600 concentrators and in how many different concentrators the company's subscriptions are connected. The business group in which the company belong is also examined. For example in Finland there are several small independent retail stores around the country who belong to a same business group called Kesko. Sometimes the centralized business group manages the subscriptions of each company belonging to this kind of group. A small part of this analysis is presented on a Table 7 below.

Table 7 Business analysis

Company Name	Business Group	Subscriptions	Subscriptions on different sites
Company A	Group A	12	2
Company B	Group B	5	4
Company C	Group C	13	1
Company D	Group A	3	1
Company E	Group E	8	2
Company F	Group F	44	26
Company G	Group G	9	3

When the analysis in the table 7 is formed, the next step is to analyse the contract situation of the customers. If a customer has a fixed-term contract, the operator can’t terminate it in a middle of the contract period. The open ended contracts, on the other hand, offer a greater possibility to offer customers new IP based replacements. From the Table 7 it can be seen that the company F have a total of 44 subscriptions and these are connected to 26 different concentrators around the country. Now if company F has a fixed term contract that runs for example another four years, its subscriptions can’t be migrated at the current time. For that reason the whole set of 26 concentrators, where company F have subscriptions, are dropped from the list of potential targets. After these 26 concentrators are removed from the list, the analysis shown in the Table 7 needs to be updated. The reason for this is that some of the remaining companies may also have subscriptions in the removed concentrators. For example the company E originally had subscriptions on two different sites. Now if both of these sites were in the set of 26 concentrators removed from the analysis, company E wouldn’t have any subscriptions at the target area. These kinds of updates were implemented manually in the Microsoft Excel because a suitable formula was not found.

There are also other criteria of why some business subscriptions can’t be migrated in the current planning period. One simple reason is that some subscriptions still create a lot of revenue. That is why the charging and billing information about each subscription is obtained from the databases. These billing lines are then combined so that the total billing of each concentrator is formed. The total income of the concentrator is then compared with different costs created by the concentrator. This simple comparison helps to identify the concentrators that are creating losses so that they can be prioritized in migration. The analysis process also needs to take into account the total value of the customer. Some customers may have subscriptions that do not create any profits for the operator but the total revenue obtained from the customer is still highly profitable. In order to maintain customer satisfaction in these situations, the less profitable subscriptions are left alone until a suitable package for the customer is implemented.

The next step is to determine what kinds of devices there are on a concentrator area. For example if there is already a TDMoP device on the same site as the concentrator, the emulation solution can be implemented more easily and it would create less costs. If there is a fibre connection to the analysed site, then a wide variety of IP-solutions is available to the customers. The 3G or 4G availability on the target area is also examined. If the concentrator area doesn’t have necessary devices to provide the replacing solutions, the migration must be performed on a later time. This depends on the investment schedule. In the analysis process, a separate section needs to be created, where the release schedule of the replacing solution and its availability are monitored.

The different factors described above are assessed and used to eliminate the targets from the buffer of 600 concentrators. There are quite a lot of important customers in the network and the business side of the operator can’t eliminate them all from the list. That is why a specific scoring mechanism was created to determine the most potential targets. In the final migration list there are still over 150 concentrators. The reason for this is that during the migration new information can arise, which could delay the migration in some areas. A lot of new information about the situation is obtained during the active migration process when the customers are contacted and their needs are identified.

9. Conclusions

This paper describes the different factors related to the service migration from the TDM based networks to an all IP environment. When the migration project started, the amount of customers connected to a device was the main driver in determining the yearly migration targets. That lead to situations where there were many devices where most of the customers were migrated successfully but one or two connections remained that couldn't be migrated. These were the connections that belonged to some important corporation or connections that were used in a service that didn't have suitable necessary replacing solution available. It was understood that further analyses in identifying the potential targets were needed. The year 2011 was the first year when a comprehensive analysis was done and the results were implemented in 2012.

The monitoring and predicting tool, described in chapter 7, main advantage has been in providing information of what is happening in the network. This information is essential in future planning and its overall use will be clarified in the future. The analysis process to identify the most optimal migration targets has clearly improved the migration process. The migration rate has significantly improved compared to the situation before the analysis process. For example the concentrator dismantling rate was 21% below the forecast in 2011 but currently in 2012 when the target analyses were first time implemented, the current speed is 30% above the forecasted rate. This clearly indicates that set goals can be achieved in 2012.

Service migration is a long-term project that will last many years. The analysis process was started with a little or with zero previous experience. That is why it is predictable that the process continues to improve during the project lifetime. This leads to a need to have resources and employees dedicated to the project who knows what has been done before and identify what needs to be improved.

The main concern for the future planning is the availably of software resources. At this point Microsoft Excel is used to implement the analyses. It is predicted that the amount of yearly migrated subscriptions grows larger later in this project and they also require more processing.

Storing of data is also a problem with Excel. In the present time Excel creates limits to the scope of the analyses. For these reasons a separate, more advanced analysis tool will be developed in the future to handle the growing data amounts. The data gathered for analyses lies around separate different databases and could come in many forms. For example some network device information is hugely different compared to the revenue information of businesses. The ideal solution for the future is to determine a separate data model that contains all the data required in migration analyses and have a tool able to process this data model. This tool would also have a separate storage where the previous actions and results are recorded.

The design and planning of this analysis tool has already started when writing this thesis. The goal is to have a tool that works as a migration database and at the same time has the sufficient analysis capabilities. In an ideal situation this tool would replace Excel altogether in the near future.

References

[1] P. Gnanasivam, *Telecommunication Switching and Networks, New Age International Publishers, 2006, ISBN: 81-224-2349-3*

[2] Jonathan Davidson, Brian Gracely, James Peters, *Overview of the PSTN and Comparisons to Voice over IP, Cisco Press, 2001*

[3] Gerald L. Hopkins, *The ISDN Literacy Book, Addison Wesley, 1995*
ISBN-13: 978-0201629798

[4] ITU-T, *Recommendation Y.2001: General Overview of NGN, 2004*

[5] Keith Knightson, *Basic NGN Architecture Principles & Issues, ITU-T Workshop on NGN (jointly organized with IETF) Geneva, 1-2 May 2005, 2005*

[6] Roger L. Freeman, *Telecommunication System Engineering, Wiley-Interscience, 2004, ISBN 0-471-45133-9*

[7] ITU-T, *Recommendation G.704: Synchronous frame structures used at 1544, 6312, 2048, 8448 and 44 736 kbit/s hierarchical levels, ITU-T, 1998*

[8] Valdar Andy, *Understanding Telecommunications Networks, The Institution of Engineering and Technology, London, United Kingdom, 2006, ISBN 0-86341-362-5*

[9] Anttalainen Tarmo, *Telecommunications Network Engineering, Artech House, 2003,*
ISBN 1-58053-500-3

[10] Ericsson & Telia, *Understanding Telecommunications 1, Studentlitteratur, 2001, ISBN 91-44-00212-2*
[11] Lydia Partizale, David T Britt et al, *TCP/IP Tutorial and Technical Overview 8th edition, IBM, 2006*

[12] Information Sciences Institute University of Southern California, *Transmission Control Protocol, RFC793, IETF, 1981*

[13] J. Mahdavi, S. Floyd, A. Romanow, PSC, LBNL, M. Mathis, Sun Microsystems, *TCP Selective Acknowledgment Options, RFC2018, IETF, 1996*

[14] T. Socolofsky & C Kale, *A TCP/IP Tutorial, RFC1180, IETF, 1991*

[15] Cisco, *Business Case: Token Ring-to-Ethernet Migration, 2000*

[16] Charles E. Spurgeon, *Ethernet: The Definitive Guide, O'Reilly, 2000, ISBN: 978-1-56592-660-8.*

[17] Metro Ethernet Forum, *Ethernet Services Definitions – Phase 2, 2008*
URL: http://metroethernetforum.org/PDF_Documents/MEF6-1.pdf

[18] E. Rosen, A Viswanathan, R. Callon, *Multiprotocol Label Switching Architecture, RFC3031, IETF, 2001*

[19] ITU-T, *Recommendation G.694.1: Spectral grids for WDM applications: DWDM frequency grid, 2002*

[20] ITU-T, *Recommendation G.694.2: Spectral grids for WDM applications: CWDM Wavelength grid, 2003*

[21] A. Vainshtein, YJ. Stein, *Structure-Agnostic Time Division Multiplexing (TDM) over Packet (SAToP), RFC4553, IETF, 2006*

[22] A. Vainshtein, I. Sasson, E. Metz, T. Frost, P. Pate, *Structure-Aware Time Division Multiplexed (TDM) Circuit Emulation Service over Packet Switched Network (CESoPSN), RFC5086, IETF, 2007*

[23] Ismail Dalgic, Michael Borella, Rick Dean, Jacek Grabiec, Jerry Mahler, Guido Schuster, and Ikhlaq Sidhu, *True Number Portability and Advanced Call Screening in a SIP-Based IP Telephony System, IEEE Communications Magazine, July 1999*

[24] Vaiva Lazauskaite, *Developments of Next Generation Networks (NGN): country case studies, ITU, 2009*

[25] Pekka Väisänen, *Quadruple Play - Promise of Bundling, Aalto University Telecommunications Forum, 2010*

[26] Agbinya I. Johnson, *IP Communications and Services for NGN, Auerbach Publications, 2010, ISBN: 978-1-4200-7090-3*

[27] Nigel Seel, *Business Strategies for the Next-Generation Network, Auerbach Publications, 2007, ISBN-13: 978-0-8493-8035-8*

[28] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler, *SIP: Session Initiation Protocol, RFC 3261, IETF, 2002*

[29] Randy Lawson, *Internet TV Subscribers to Rise by More Than 50 Percent in 2009, iSuppli, 2009, [referenced 18.3.2012]*

[30] Broadband forum press release, Global Broadband Hits 484 Million Subscriber Mark as More Than 14 Million Lines Are Added In Q1 2010, Broadband forum, 2010, [referenced 18.3.2012]
URL: http://www.broadband-forum.org/news/download/pressreleeases/2010/Q110stats.pdf

[31] Finnish Communications Regulatory Authority (FICORA), *Half-year report of the communications market 2011, 2011*

[32] Finnish Communications Regulatory Authority (FICORA), *Market Review 2007, 2007*

[33] CIA, *The World Factbook ISSN 1553-8133*

[34] The Fiber Optic Association, *How Fast Is Data Traffic Growing?, 2011*

[35] Cisco, *Cisco Visual Networking Index: Forecast and Methodology, 2010–2015, 2011*

[36] K. Rezabeigi, A. Vafaei, N. Movahhedinia, *A Web Services based Architecture for NGN Services Delivery, World Academy of Science, Engineering and Technology 43 2008, 2008,*

[37] David Mongeau & Govinda Rajan, *Optimizing Costs of Migration to NGNs, Networks 2008 13th International Symposium 2 October 2008, 2008*

[38] Arvind Kumar, *Learning's from NGN Migration Study Group, Workshop on NGN Regulation and Migration Strategies New Delhi October 13 & 15 2010, 2010*

[39] Ahmad A.Almughaless & Ali M Alsaih., *Optimum Migration Scenario From PSTN to NGN, Communication Systems, Networks and Applications (ICCSNA) 2010 Second International Conference on, IEEE, 2010*

[40] Chris De' Courey Bower, Wouter Franx & Manfred Hammer, *Overlay NGN Migration Approach to Deliver Multimedia Services, Telecommunications Network Strategy and Planning Symposium. NETWORKS 2004, 11th International, 2004*

[41] Nokia Siemens Networks, *SURPASS hiE 9200 product flyer, NSN, 2008*